

# **ELECTRONIC FUNDAMENTALS**

## **SERVICE PRACTICES 34**

### **HIGH-FIDELITY**

- 34-1. Fundamentals Of Sound**
- 34-2. Acoustics**
- 34-3. Record Recording Characteristics**
- 34-4. Types Of Pick-Up Cartridges And Styli**
- 34-5. Pre-Amplification And Equalization**
- 34-6. Tuners**
- 34-7. Power Amplifiers**
- 34-8. High-Fidelity Speakers And Crossover Networks**
- 34-9. Speaker Enclosures**
- 34-10. High-Fidelity Amplifier Measurements**



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**A SERVICE OF RADIO CORPORATION OF AMERICA**  
**HOME STUDY SCHOOL**  
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# Service Practices 34

## 34-1. FUNDAMENTALS OF SOUND

When servicing high-fidelity equipment, it is important that the serviceman be capable of judging the quality of sound. The customer's disapproval of the instrument may be due to the irritating sounds he hears or the pleasant tones he does not hear. Thus, it is not only necessary to evaluate the customer's complaint, but be able to interpret what you hear in terms of circuit defects. When the customer smells smoke, or hears nothing, your problem is a simple one. However, when the customer is complaining of not hearing the resin on the bow from

Beethoven's Violin Concerto in D, the technician is sometimes at his wit's end trying to determine exactly what the customer has in mind.

This lesson will cover the major components in a high-fidelity system and explain their limitations. With this background the technician can judge a high-fidelity system's performance and make a repair.

**Nature of Sound Waves** — Before going further in the subject of high-fidelity, it may be well to consider the nature of sound waves. The earth is surrounded by a deep ocean of air and we live on the bottom of this ocean.

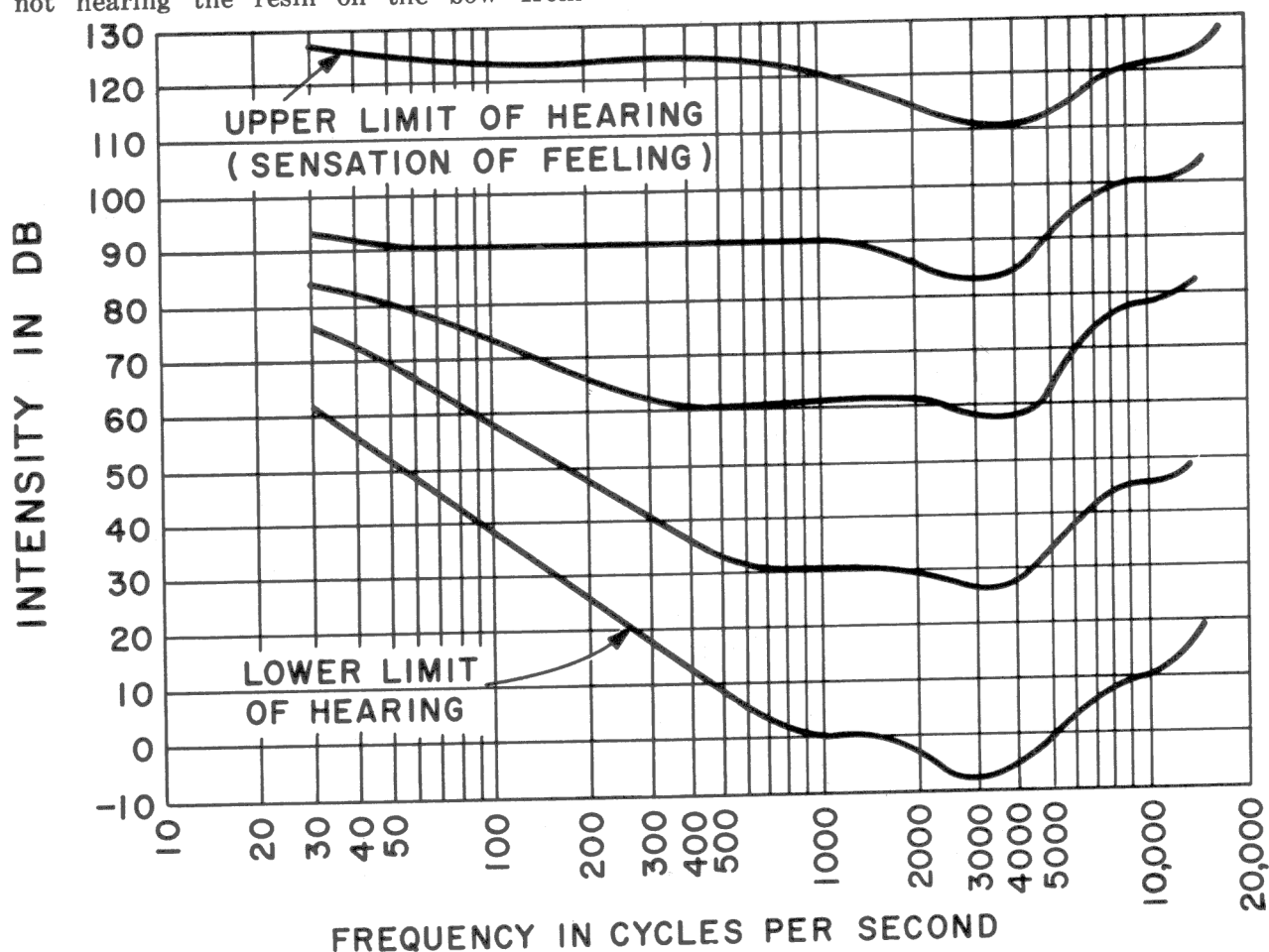
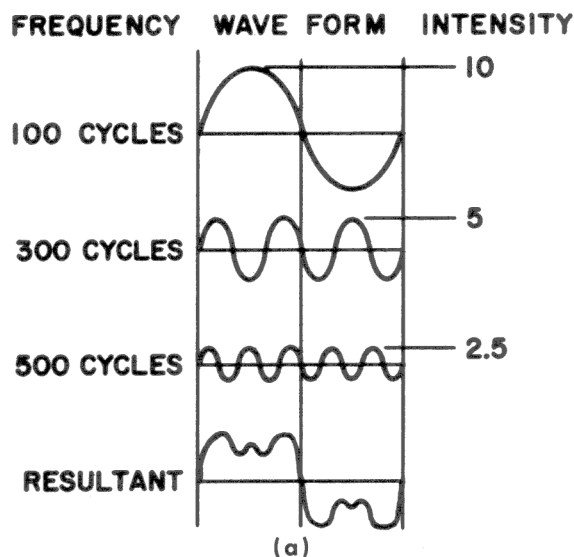
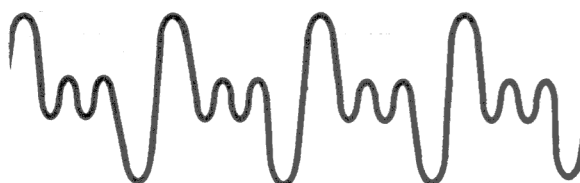


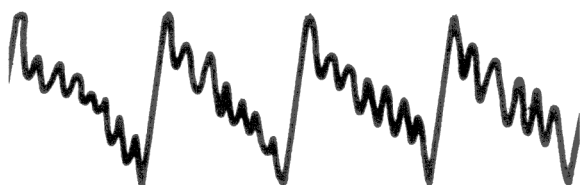
FIGURE 1



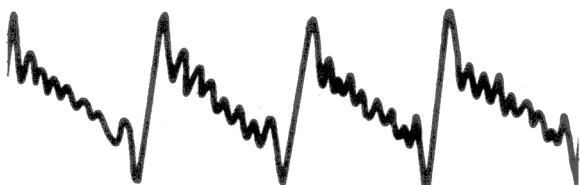
### MIDDLE C



### CELLO ORGAN PIPE



### TROMBONE ORGAN PIPE



### PIANO



### "AH" SUNG

FIGURE 2(b)

The miles of air above us have a certain weight or pressure which is approximately fifteen pounds per square inch at sea level. If you were to clap your hands you would force the air between your hands outwardly in all directions in the form of sound waves traveling at a constant velocity. These sound waves could be intercepted at a distant point if a detecting device (such as the human ear) were present. Sound is then simply a sensation produced through the ear when vibrations are set up in the surrounding air. These sound waves have two important characteristics; frequency, which determines the pitch of the sound, and amplitude, which determines the intensity of the sound.

The intensity of the sound wave determines the amplitude of the vibration within the ear. The response of the ear to sounds of different intensities is proportional to the logarithm of the intensity, rather than directly to the intensity. In other words, if there were three sounds with intensities of 1, 10 and 100 units, the ear would pick-up the same relative difference in intensity between the 1 and 10 unit sounds and the 10 and 100 unit sounds, since the ratios are 10 to 1 in each case.

The response of the ear varies both with the frequency and intensity of sound. Figure 1 shows the response of the average ear at different frequencies and intensity levels. These curves show that the response is reasonably flat at high intensity levels but falls off considerably at certain frequencies at low intensity levels. This is the main reason we use compensated volume controls, or loudness controls in most reproducing instruments (high-fidelity and otherwise). Each curve in Figure 1 is based on equal loudness with the bottom curve representing the levels where the sound is just perceptible, and the top curve representing the level above where the sound becomes feeling.

**Quality of Sound** — Figure 2(a) shows various intensities caused by single frequency notes. This wave is produced by the fundamental wave indicated at the top, plus the third and fifth harmonics. The wave at the top is one cycle of a 100 cycle per second wave having an amplitude of ten units. If we add to this wave the third harmonic, which is 300 cycles having an amplitude of five units, and the fifth harmonic which is 500 cycles

having an amplitude of two and one half units, the resultant wave form shown at the bottom would be obtained. It is the harmonic frequencies that give the characteristic quality to a particular note. However, such sine wave notes are not commonly produced by mechanical means. Most sounds contain not only one basic frequency, but harmonics of that frequency as well. For instance, in Figure 2(b) a sound wave is shown which most nearly conforms to the sound waves made by musical instruments. For example, middle "C" on a piano sounds different than middle "C" on a cello organpipe. The difference heard is dependent on the number and amount of harmonics (overtones) produced by each instrument. The same rule applies for sounds that are not musical notes. For instance, one person's voice is different from another person's voice because of the overtones it produces. Our voices are produced by the vibration of the vocal chords and the resonance of the mouth and throat cavities. Some people have long necks or vocal chords making them bass singers, while others have short vocal chords making them tenor singers.

Most high-quality audio systems installed in homes are used mainly for the reproduction of music from the excellent present-day recordings, and many musical programs broadcast through the media of AM, FM and TV. The qualities necessary for obtaining listening satisfaction and pleasure receive much consideration during the development of the audio system. Usually after a wide range output free from disturbing noise and distortion has been achieved, the designer will try to bring out that quality of sound which will make it seem alive and realistic. This is a quality difficult to describe, but it is the aliveness or presence heard and felt which determines the difference between listening to a live performance or listening to a reproduction over a high-quality sound system. The ultimate goal would be to have the sound so realistic that if you were to close your eyes you could not tell the difference between the sounds coming from the loud-speaker system and the actual live performance.

Another important consideration in determining a high-quality system is *listeners*

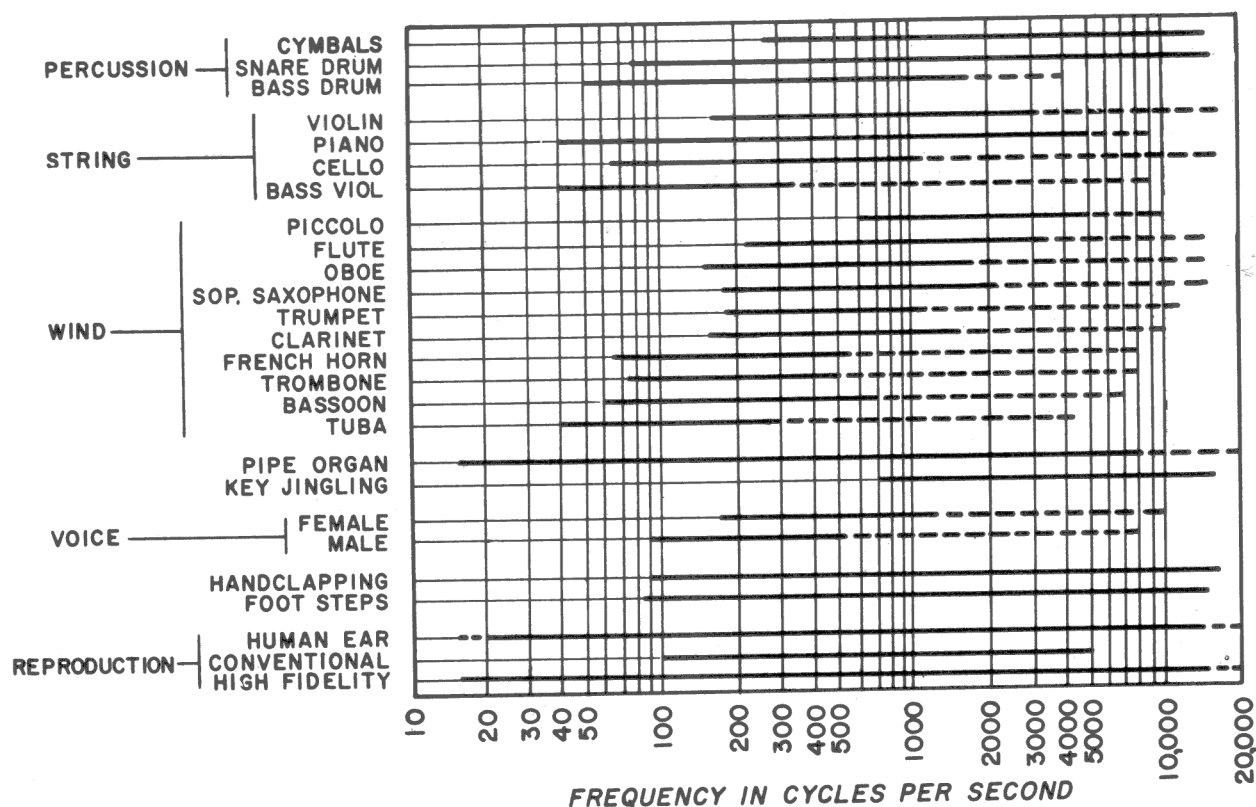


FIGURE 3

*fatigue*. This is caused by a system producing sharp or shrill tones or booming bass that makes one want to turn the audio system off after a short listening period. A good check for listeners fatigue would be to turn the audio system on and listen for a period of time. If you enjoy the sounds coming from the loudspeaker without fatigue you can be assured that the quality of sound is good. But on the other hand, if you can't wait to turn the "thing" off, you have what is known as listeners fatigue.

#### Over-Tone Range of Various Instruments—

Figure 3 shows the range of various musical instruments. The solid lines indicate the fundamental frequencies and the dotted lines indicate the overtones. Several interesting things can be seen in this figure. Note, for example, the range of the snare drum which is a percussion instrument compared to the violin which belongs to the string section. The extended frequency range of the snare drum is caused by several wires on the under-

side of the drum head that vibrate against the head to produce overtones. The sound of the bass drum is also high although the fundamental frequency is only a dull thud.

It can also be seen from Figure 3 that the frequency range of the female voice extends from 170 cps to 10,000 cps as compared to the male voice which extends from 100 cps to 8,000 cps. The sound produced by hand-clapping, although not a musical note, ranges from 100 cps to 16,000 cps which is beyond the range of many people. Only a few instruments have fundamental frequencies exceeding 5,000 cps, but their tonal ranges are often several times 5,000 cps because of their high harmonics.

### 34-2. ACOUSTICS

#### Frequency and Intensity Ranges of Sound—

The sounds produced by a symphony orchestra in a large hall cover a wide range of frequencies and intensities that can not be accurately reproduced in the home. As previously mentioned the response of the ear is

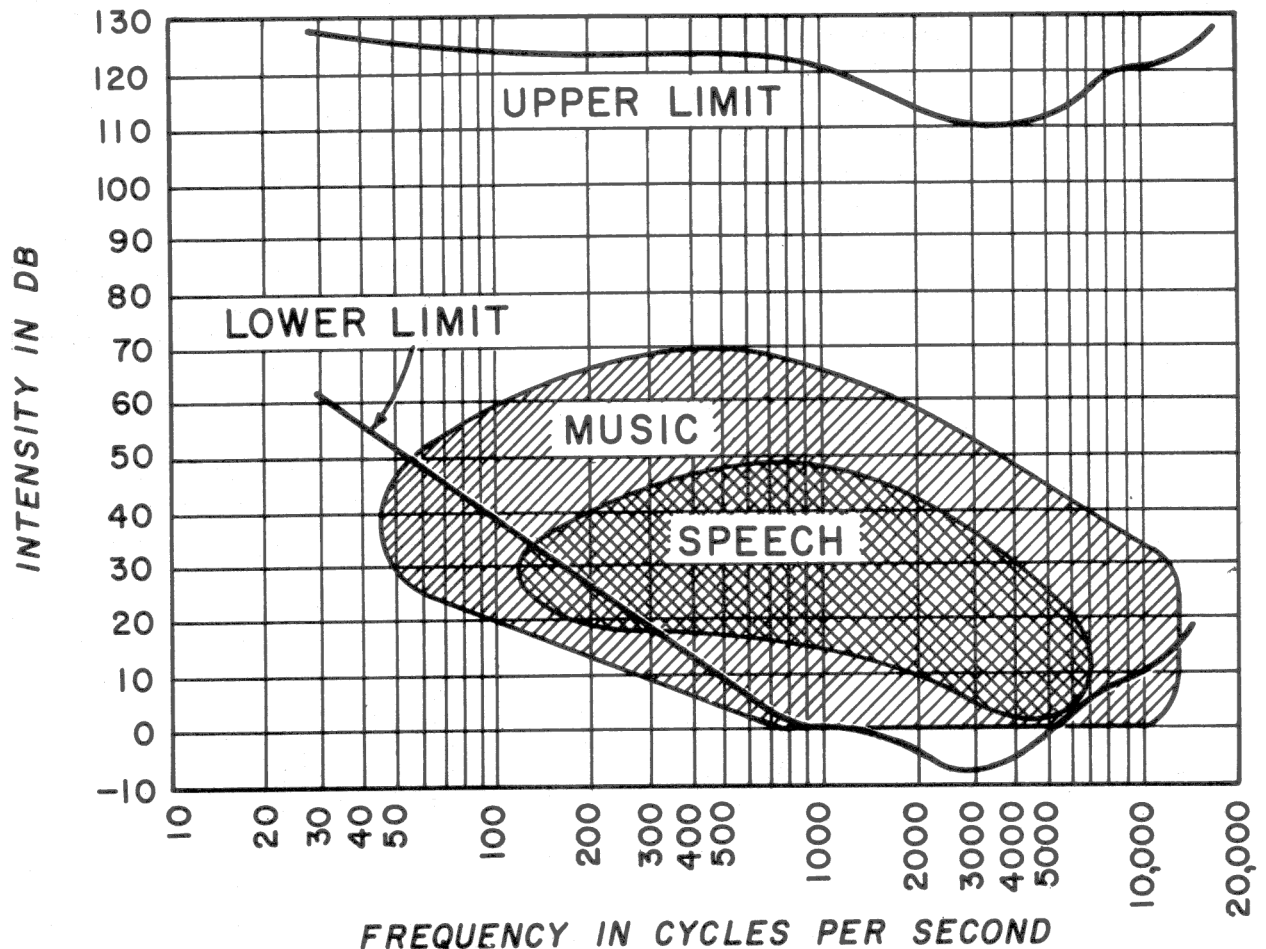


FIGURE 4

not flat at all frequencies and volume levels. Figure 4 shows the comparison between music and speech frequencies and their volume levels. You will note that music covers a much wider range of frequencies and intensities than speech. Both the upper and lower limits of the ear response are shown.

In order for an instrument to be called high-fidelity, the frequency range must be broad enough to cover the low frequencies as well as the middle and high frequencies and most of their harmonics. An acceptable high-fidelity system should be able to reproduce frequencies from 70 cps to 10,000 cps. However, you will find that the majority of the better equipment on the market today will faithfully reproduce any frequency from 30-40 cps to 20,000 cps. The upper limit of 20,000 cps is necessary since each instrument has a characteristic tonal quality which is made up of the fundamental frequency plus the harmonics. For example, an instrument such as the organ is capable of producing a fundamental frequency of 5,000 cps. If the high-fidelity system is not capable of reproducing the associated harmonics the organ will no longer sound like an organ.

**Reverberation** — Reverberation is simply echoed sound in a room due to reflections from walls, ceiling, floor, furniture, and etc. In other words, sound waves reach the listener through both direct and indirect paths. The sounds following the direct path will reach the listener first. The reflected sounds, which have followed longer paths, combine with the direct sounds, as shown in Figure 5. By interrupting the sound source, the reflections continue for a short period of time depending on their reflected path length. The short period of time between the interruption of the sound source and the end of the reflected sound is called the reverberation period. Because of reverberation, the location of a high-fidelity instrument in a room is of the utmost importance. To obtain the best results, the instrument should be placed in the room so that the wall which is perpendicular to the sound waves leaving the speaker is as far away as possible, or heavily draped.

### 34-3. RECORD RECORDING CHARACTERISTICS

A phonograph record is simply a flat disc with numerous spiral grooves. The manner

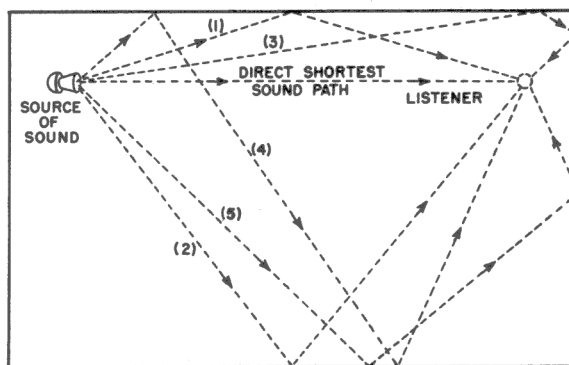


FIGURE 5

in which the spiral grooves are altered as the master record is cut is known as the recording characteristic.

The device that cuts the record is called the cutting head. This head resembles the tone arm and pick-up of a normal record player. The major difference is the fact that it is usually heavier and uses a heated stylus to smoothly cut a spiral groove into the blank master record.

There are basically two types of cutting heads, namely the magnetic and crystal. The magnetic type is most widely used, therefore, the following discussion will be based on this type.

When an audio signal is fed to the cutting head the stylus moves in a *lateral* direction forming a "wiggly" groove as the head spirals the record. The amount of this *lateral* movement varies *inversely* with frequency. In other words, as the frequency *increases* the *lateral* movement of the stylus *decreases*, assuming a constant amplitude signal, at all frequencies, is fed to the cutting head. This groove modulation verses frequency is illustrated in Figure 6(a).

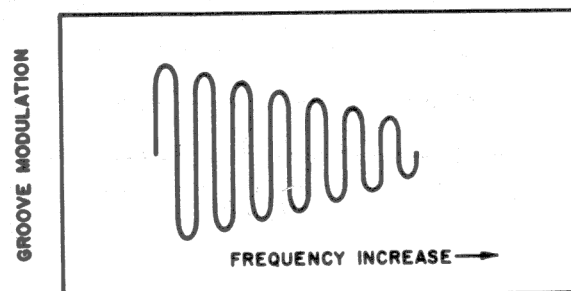


FIGURE 6(a)

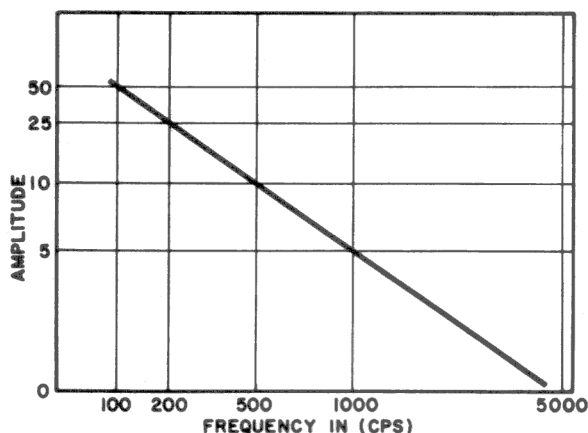


FIGURE 6(b)

It can be seen from the graph in Figure 6(b) that as frequency is doubled the amplitude (or intensity) is decreased by one-half. This recording characteristic is known as "constant velocity." The formula for "constant velocity" is  $2\pi fa$ , "f" being the frequency recorded and "a" being the amplitude. The product of frequency and amplitude must always equal the same number if the velocity is to remain constant. For example, let us assume the *lateral* velocity of a stylus to be a constant 3"/sec. From this constant the amplitude for various frequencies can be calculated. By dropping the  $2\pi$  from the formula we have:

$$\text{Constant Velocity} = fa$$

$$3"/\text{sec.} = fa$$

$$3"/\text{sec.} = 4,000 \text{ cps} \times "a"; a = .00075"$$

$$3"/\text{sec.} = 1,000 \text{ cps} \times "a"; a = .003"$$

$$3"/\text{sec.} = 500 \text{ cps} \times "a"; a = .006"$$

$$3"/\text{sec.} = 125 \text{ cps} \times "a"; a = .024"$$

From the above we can conclude the following:

If a constant amplitude signal is applied to the cutting-head the groove modulation will vary inversely with frequency. As can be seen from the formula  $2\pi fa$ , as the frequency *increases* the *amplitude* of cut or *groove modulation decreases*. As frequency *decreases* the *amplitude* of cut *increases*.

Groove modulation is limited by the number of grooves per inch on the record. In order to obtain reasonable playing time on a record, the number of grooves per inch varies from 96-275. When using 100 grooves per

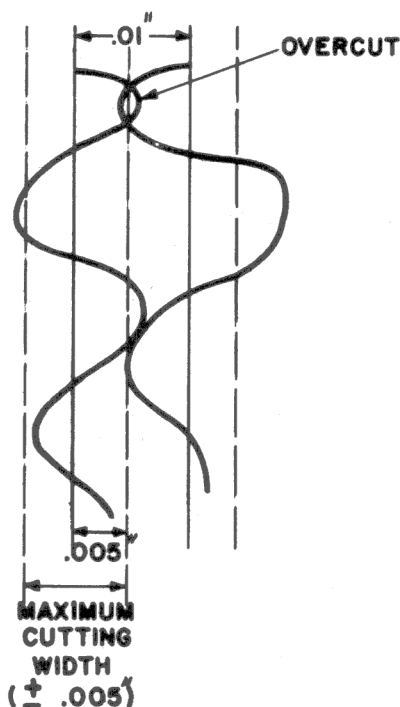


FIGURE 7

inch the spacing from one groove center to the next would be 1/100th of an inch or .01 inches. If the lateral swing of the stylus exceeds this, there will be what is commonly called an *over-cut* as shown in Figure 7. Over-cuts result in the stylus crossing from one groove to another. To prevent over-cuts the amplitude of frequencies below 500 cps should be limited to .005 inches.

Until now we have been speaking of "constant velocity" recording, that is, feeding a constant amplitude signal into the cutting head with the result that groove modulation varies inversely with frequency. To prevent the frequencies below 500 cps from over-cutting the record, the signal amplitude fed to the cutting head is reduced as shown in Figure 8(a). Thus, the groove modulation will be *flat* or *constant* from 20-500 cps. However, groove modulation will decrease rapidly as frequency increases above 500 cps. At 15,000 cps the inherent record *scratch* or *noise* over-rides the groove modulation with the result of noisy high-frequency reproduction. Therefore, the high-frequency signal amplitude is *pre-emphasized* or *increased* before it is applied to the cutting head as seen in Figure 8(b). This results in a relatively constant *amplitude* or groove mod-

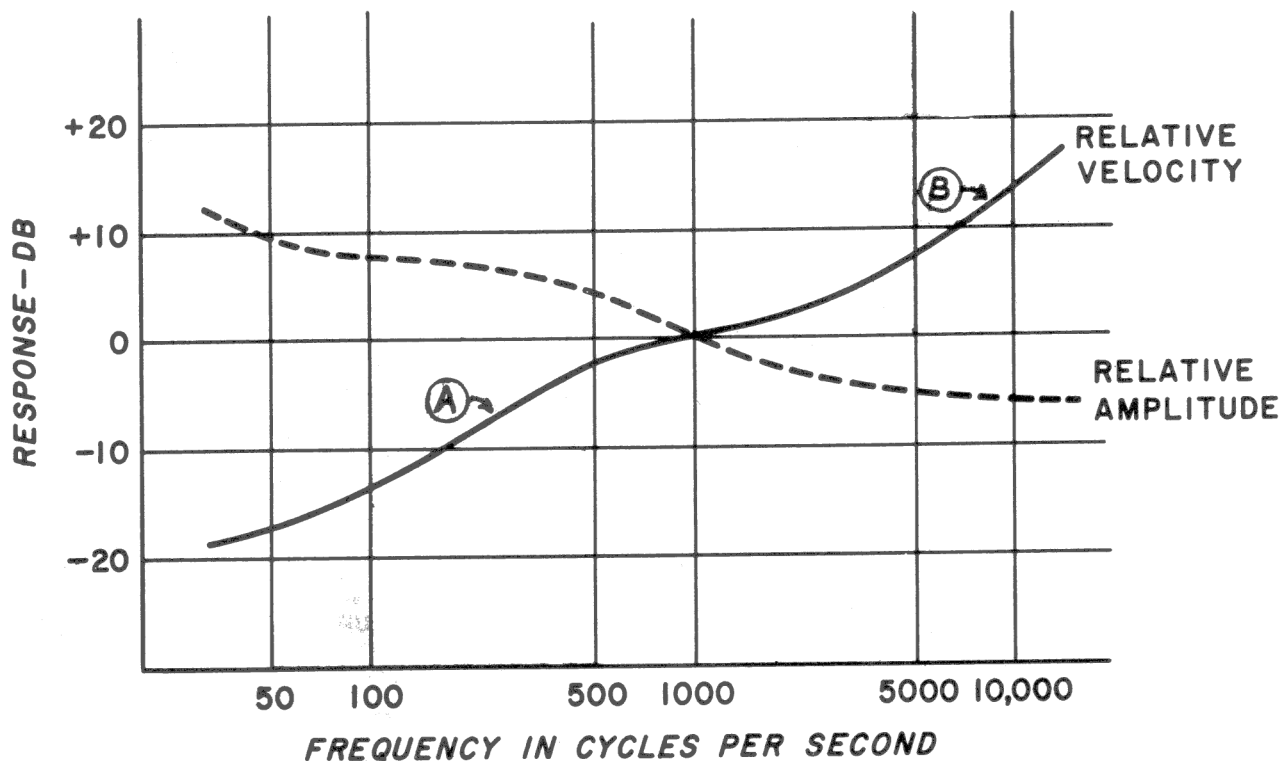


FIGURE 8

ulation recording characteristic as shown by the dotted line in Figure 8. The overall deemphasis or decrease on lows and pre-emphasis on highs is shown by the solid line in Figure 8.

#### 34.4. TYPES OF PICK-UP CARTRIDGES AND STYLI

**The Styli** — The stylus rides in the record grooves and produces small mechanical impulses that travel from the stylus to the pick-up cartridge. Here they are converted to electrical impulses which are fed to the amplifying system. The stylus was at one time called the phonograph needle, but it no longer takes the form of a needle so it is now called the stylus. There are various types of styli in use today, each one having its own characteristic. The most common are the osmium, sapphire and diamond.

A standard 78 rpm record has about 100 grooves per inch, and a  $33\frac{1}{3}$  rpm micro-groove record has 225-275 grooves per inch. The ratio of grooves per inch is approximately 3 to 1. Therefore, the ratio of groove diameters is 3 to 1 with 78 rpm grooves being the larger of the two. The difference in groove diameters is the reason for using two

different size styli. The 78 rpm records require a 3 mill stylus, whereas  $33\frac{1}{3}$  rpm records require a 1 mill stylus. Figure 9 shows a  $33\frac{1}{3}$  rpm record stylus tip and how it fits in the groove. It can be seen from this draw-

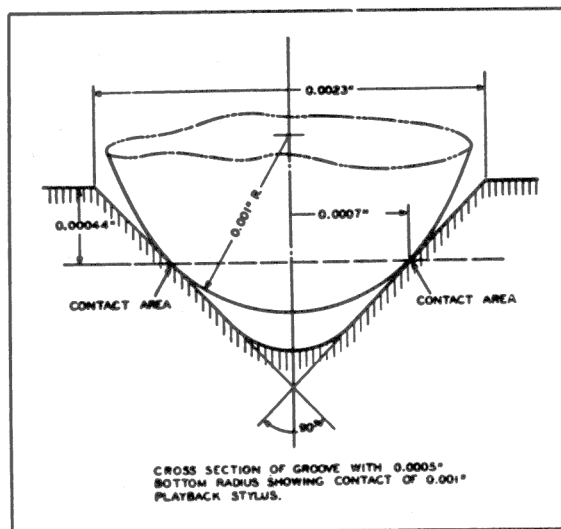


FIGURE 9



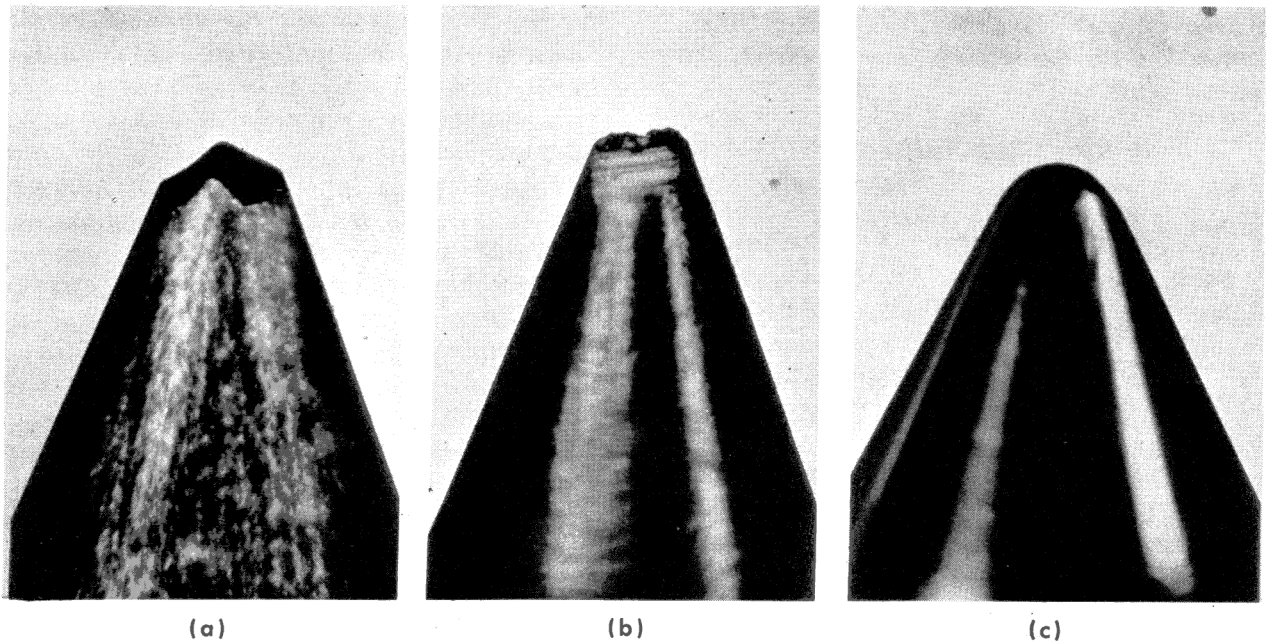


FIGURE 10

ing that the stylus touches the record groove at only two microscopic points. Although the pick-up and tone arm may be very light (approximately 6 grams) the pressure on the stylus at these two points is rated in tons per square inch. This pressure is the primary reason for stylus wear. The drawing in Figure 10 shows a comparison of the osmium, sapphire and diamond styli after 72 hours of play on new micro-groove records. In Figure 10, (A) is the osmium, (B) is the sapphire and (C) is the diamond. Both the osmium and sapphire, as can be seen, show flat spots worn on the sides of the stylus. The diamond shows no sign of wear. From this figure it can be seen that the diamond stylus will out-wear the two other types.

The primary concern with the stylus is replacement. The following methods can be used to check a suspected worn stylus:

1. By examining the stylus under a microscope. If the flat spot on the stylus tip is greater than one-half the diameter, the stylus should be replaced (some diamond styli may be resharpened).
2. By direct replacement, provided the part is readily available.
3. By listening to determine if the recorded music has lost its fidelity from excessive distortion caused by a worn stylus. (Use record known to be good).

The preferred method for checking a stylus is by close inspection under a microscope.

However, any one of the previous methods can be used as a means of checking a stylus.

**Pick-Up Cartridges** — Pick-ups can be classified into two general groups. The high-impedance type and low-impedance type. The pick-ups that fall into the high-impedance group are the crystal and condenser. The *crystal pick-up* is most widely used for the reproduction of sound from the phonograph record. The crystal pick-up employs the Rochelle salt crystal element which is caused to twist due to the movement of the stylus and thus, generates a voltage which is *directly proportional* to the amplitude of the record groove modulation.

The construction of a crystal is shown in Figure 11 (a). The crystal is supported in the cartridge so that one end is held securely.

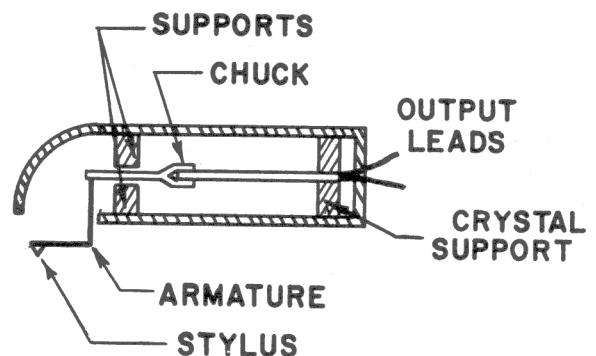


FIGURE 11(a)

The other end has a flexible support which allows the crystal to be twisted by the fluctuating stylus.

### EQUALIZED RESPONSE

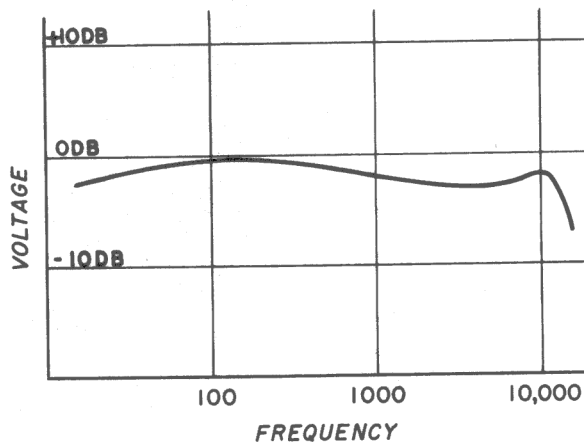


FIGURE 11(b)

The frequency response of the crystal pick-up is shown in Figure 11(b). Note that the curve is flat within  $\pm 5$ db from 50 cps to 10,000 cps and then falls off rapidly above 10 kc. This high-frequency response is the primary reason why crystal pick-ups have only a limited use in modern high-fidelity systems.

The *condenser pick-up* also belongs to the high-impedance group. The operation of the condenser pick-up is similar to the condenser microphone. One capacitor plate is mounted at the end of the pick-up nearest the stylus, while another capacitor plate is mounted toward the rear of the pick-up. The plate nearest the stylus is moveable and the plate at the rear uses a fixed mount. As the stylus moves laterally in the record grooves the varying capacitance generates a small signal voltage, *proportional to the change in capacitance*. The response from the condenser

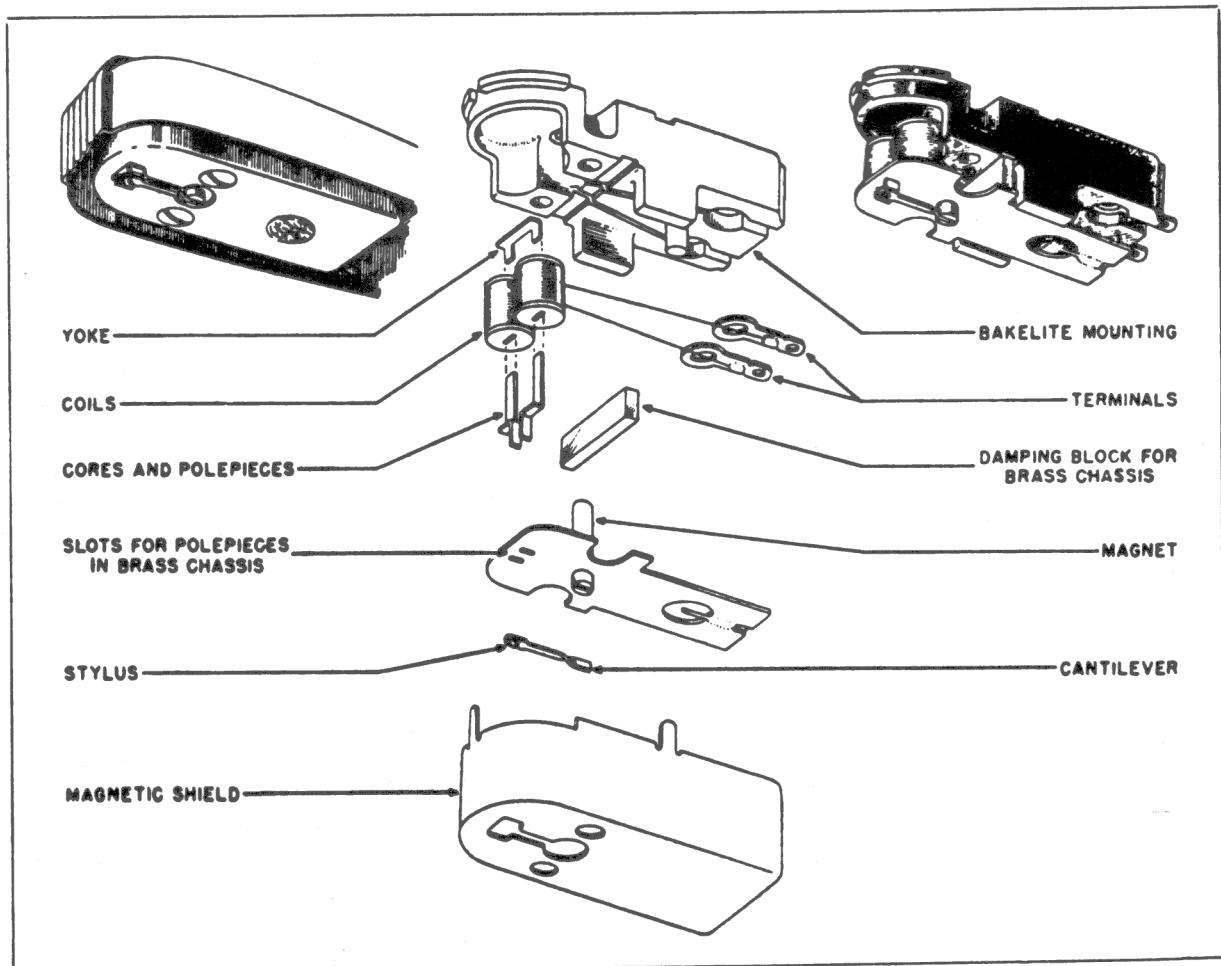


FIGURE 12

pick-up is somewhat better than the crystal pick-up, however, due to its higher cost, it has limited use.

The low-impedance pick-ups consist of the magnetic type. There are two types presently in use: (1) The variable reluctance type; (2) the moving coil type. The variable reluctance (V-R) pick-up is shown in Figure 12. Two coils electrically connected in push-pull are mounted as shown. The stylus is mounted at one end of a cantilever spring and the other end of the spring is soldered to a non-magnetic bushing which in turn is fastened to the magnet. The stylus end of the cantilever spring is centered between the two pole pieces. Thus, with *no* lateral stylus movement the magnetic flux is distributed equally between the two pole pieces. When the stylus begins its lateral movement the output voltage generated in the coils is *directly proportional to the rate of change of flux* in the two coils. The rate of change of flux is *directly proportional to the rate of change of lateral stylus movement*. Thus, the stylus responds accurately to a *constant velocity* record groove modulation. This pick-up has the

same characteristics as the constant velocity recording head. Therefore, if the record is recorded with a constant velocity characteristic the output of the V-R pick-up will be flat. The records are, however, recorded *constant amplitude* from 30-500 cps and *constant velocity* with high-frequency *pre-emphasis* from 500-15,000 cps. Thus, the output of the V-R pick-up must be equalized. The method of equalization will be covered later in this lesson.

The moving-coil pick-up generally known as the dynamic pick-up operates on the same principle as the V-R type, except that a coil is mounted on a spring toward the rear of the pick-up. Adjacent to the coil is a magnet. The stylus moves the coil which intersects the lines of flux of the magnet, producing audio frequency voltage across the coil. The output voltage is dependent on the *rate of change of lateral deflection* of the stylus in the record groove. This pick-up has a frequency response to 15,000 cps and has the same constant velocity characteristic as the V-R type. The response curve with equalization is shown in Figure 13.

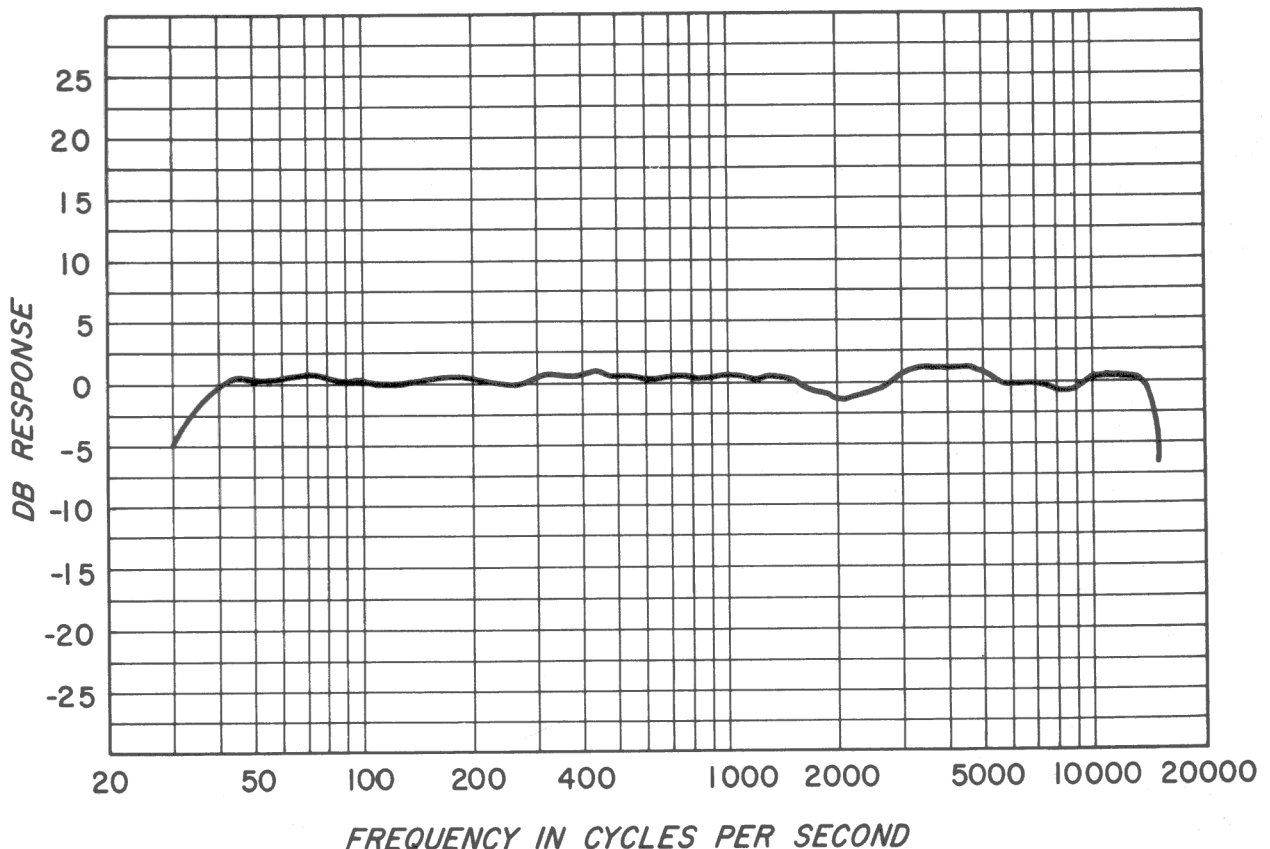


FIGURE 13

The ability of a pick-up to move freely is known as *compliance*. A pick-up assembly that moves *freely*, both vertically and laterally, has a high compliance compared to a rigid or stiff moving assembly with *low* compliance. A pick-up that has a stiff movement creates friction at the two points where the stylus makes contact with the record groove. This friction is another cause of stylus and record wear.

### 34-5. PRE-AMPLIFICATION AND EQUALIZATION

A pre-amplifier contains one or more stages of amplification and is used to drive the main amplifier. There are some instruments that have the pre-amplifier, main amplifier, and power supply mounted on one chassis. The majority of the hi-fi instruments, however, use a separate chassis for the pre-amplifier, because it provides better shielding from noise and hum that might distort the low signal voltage present at the input of the pre-amplifier. If we were to insert a signal from a crystal pick-up or tuner into the main amplifier, there would be no problem from noise or hum due to the high signal to noise ratio. The voltage output of the magnetic pick-up is approximately 1/100th the signal from the crystal pick-up. Therefore, it is necessary to amplify these weak signals before they enter the main amplifier for the simple reason that any noise, interference or hum can modulate the weak signals at the amplifier. Thus, even the requirements of the pre-amplifier are critical because if any noise were to interfere with the signal at this point, it would be greatly amplified before it reached the speaker.

Up to this point in our pre-amplifier discussion we have assumed that the signals entering the pre-amplifier have a flat response. However, this is only true when a signal from a tuner, microphone, TV sound, crystal pick-ups and compensated tape recorders are fed into the pre-amplifier. The exceptions are the signals from the magnetic pick-ups and uncompensated tape recorders.

We have already discussed the recording characteristics, but now we have a signal present on the two leads from the magnetic pick-up and the big problem is do we have to boost the bass and treble—and where and how? Each recording manufacturer has his own ideas about the recording characteristic,

despite the fact that trade organizations have attempted to set a standard. Therefore, the high-fidelity instrument manufacturers must make a product that will complement all of the recording characteristics presently in use.

**Pick-Up Equalization** — The all-important job of the equalizer is to compensate for the various record recording characteristics when using a magnetic pick-up. The equalizer can be purchased separately or a combined unit with the pre-amplifier. The simplest units usually provide equalization for the following recording characteristics.

**The RCA Orthophonic** playback curve is illustrated in Figure 14 (a). This playback curve is intended for all RCA Orthophonic recordings. The response curve shows the bass boosted approximately 18 db at 30 cps extending out to 14 kc with about 18 db attenuation.

**AES (Audio Engineering Society)** playback curve is shown in Figure 14 (b). This playback characteristic is intended as a compromise for all modern recordings. The curve gives a bass boost of 20 db at 30 cps and 17 db attenuation at 15 kc. The AES playback characteristic can be used in place of the LP curve if additional bass boost is preferred. This curve can be used for new 78 and 45 rpm records with slight adjustment of the tone controls from their flat position.

**The LP** playback curve is shown in Figure 14 (c) and is generally intended for 33 $\frac{1}{3}$  rpm records having the longplaying characteristic. The playback curve is boosted 15 db at 30 cps and extends out to 15 kc with 22 db attenuation.

**The Early 78** playback curve is illustrated in Figure 14 (d). This curve is intended for records having a noticeable surface noise on the record. The response curve shows a 20 db boost at 30 cps with all frequencies above 7,500 cps sharply attenuated.

There are several equalizing positions provided on certain equalizer pre-amplifiers that we have not mentioned, however, the playback curves just described will suffice for the majority of recording characteristics, plus a slight adjustment of the tone controls from their flat position.

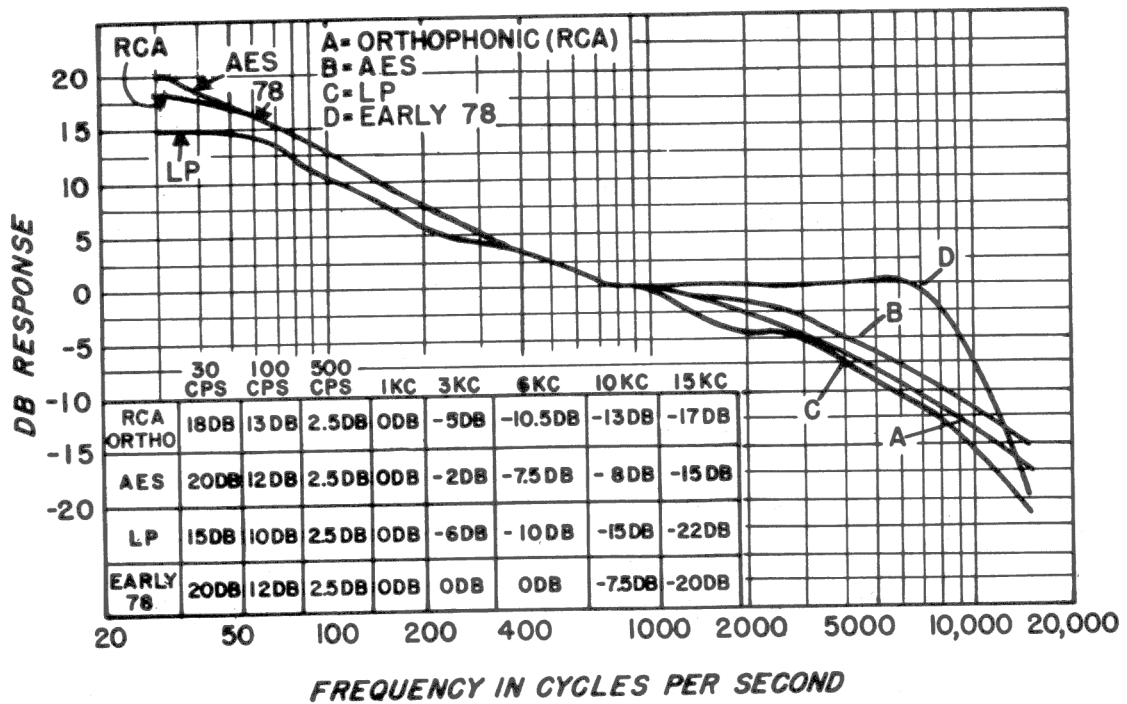


FIGURE 14

**Effect of Circuit Components**—High-fidelity circuits use components which should be familiar to you after reading the previous home study lessons. However, these components are used in circuits in a different manner. Therefore, it will be beneficial if we

review RCL circuits for a few moments before discussing high-fidelity circuitry.

First, let's look into some simple circuits in which the capacitance and inductance are connected in series parallel with a resistor. In Section A of Figure 15 two resistors are

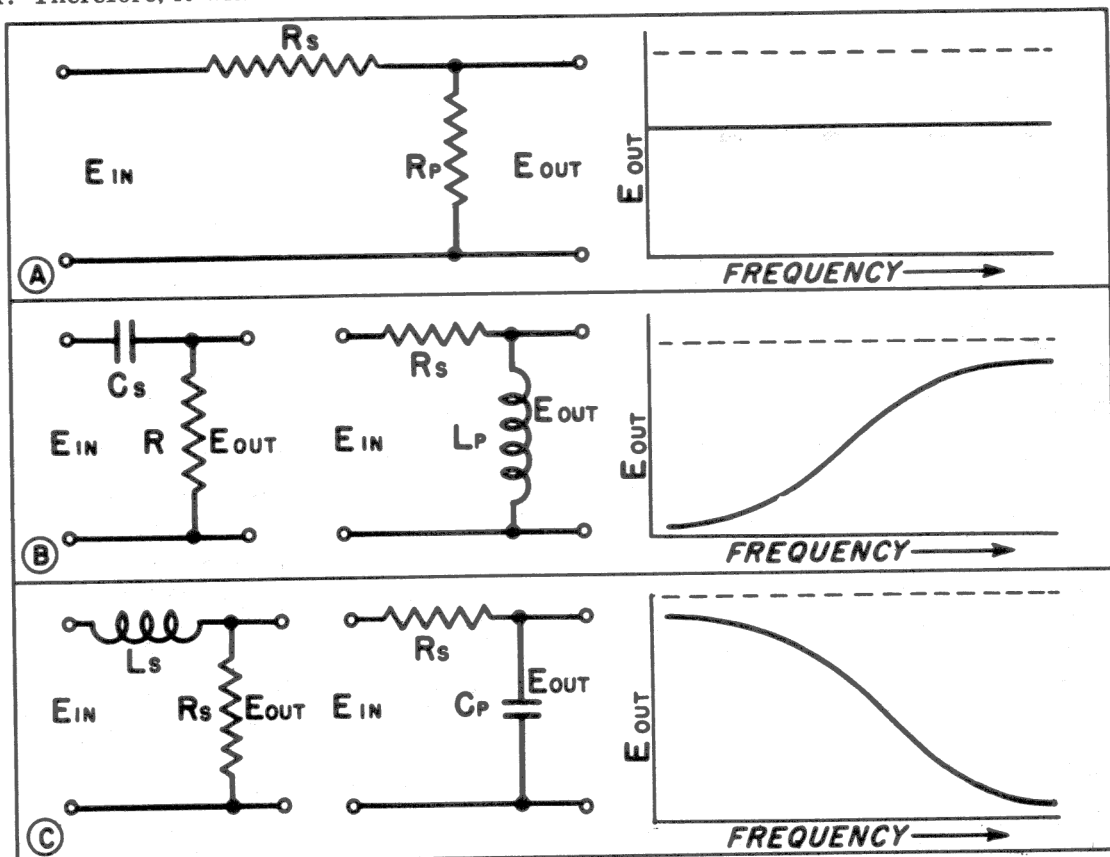


FIGURE 15

shown connected as a simple attenuator circuit. Resistors are not frequency selective and, therefore, if we apply a voltage  $E_{in}$ , represented by the dotted line, to the input network the output voltage will be attenuated as shown by the straight line and can be measured across resistor  $R_p$ . This attenuation will be governed by the voltage drop across resistor  $R_s$ .

Section B shows two circuits, one containing R/C and the other R/L. A capacitor in series with the input has the same effect on frequency as an inductor in parallel with the output. If a voltage is applied to the input, the attenuation or loss at the low frequencies is high. This type of circuit is a frequency selective device.

In Section C two circuits are shown, one with an inductor in series with the input voltage and the other with a capacitance in parallel with the output. Since the inductor offers a low impedance and the capacitor offers a high impedance to low frequencies, the output curve resembles the one shown in "C."

In each section of Figure 15 it can be seen that the resistor acts as either a loss or a load to the circuit. We can also see that in each case we have no bass boost or high-frequency boost. In Sections "B" or "C" there is a loss of either the high or low frequencies due to the inherent attenuation of the particular network.

**Tone Control Characteristics** — Figure 16 illustrates a tone control circuit which consists of a high and low frequency circuit. The treble or high-frequency circuit is composed of resistor  $R_{57}$  and three capacitors— $C_{45}$ ,  $C_{46}$  and  $C_{48}$ . The capacitive section serves as a load for the voltage divider network. As the slider on  $R_{57}$  is moved to the high frequency attenuation or "cut" position, the resistance in series with the signal is higher and the impedance at high frequencies to ground is lower; therefore, the high-frequency voltage present on the following tube grid is lower.

The bass or low-frequency control consists of capacitor  $C_{47}$  and the load resistors  $R_{54}$  and  $R_{56}$ . At the low frequency attenuation or "cut" position the impedance of  $C_{47}$  is

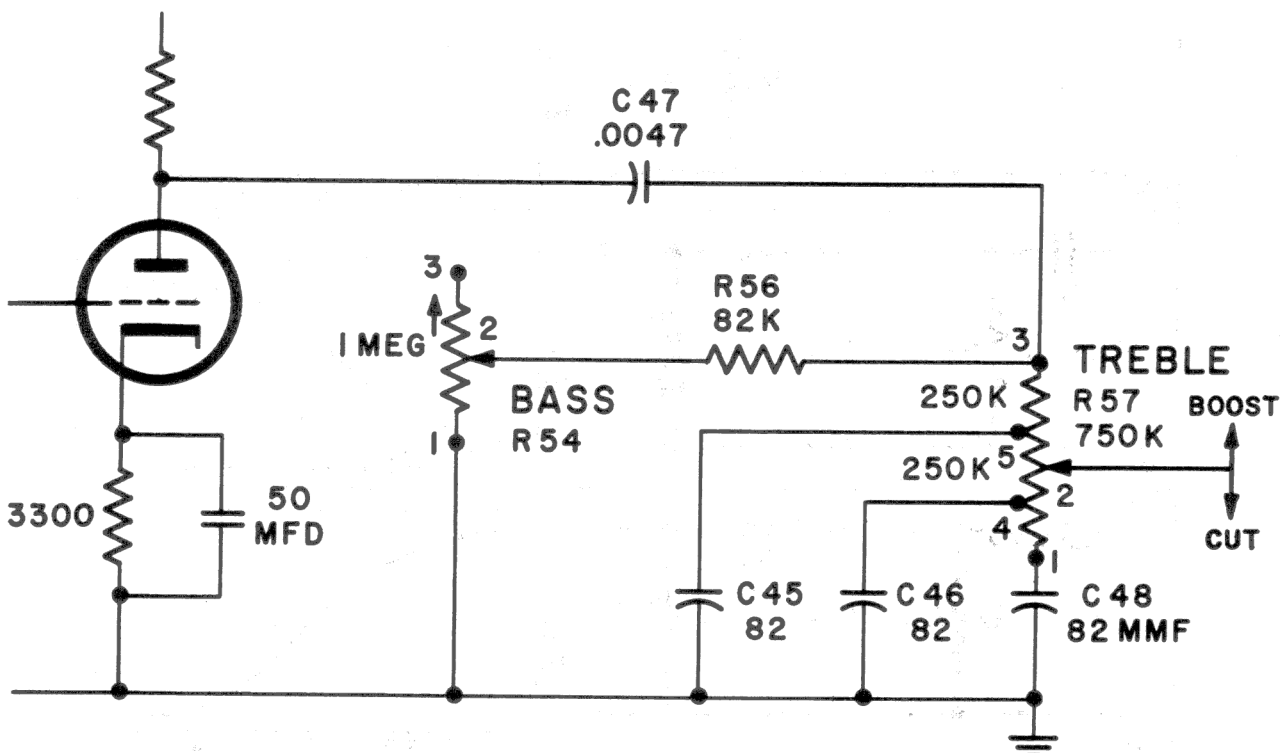


FIGURE 16

high compared to the resistive load; therefore, the low-frequency voltage on the following grid will be minimum. In most high-fidelity instruments, both the bass and treble controls should be set at mid-position. This allows for adjustment of either control to compensate for personal listening requirements, recording characteristics and acoustical qualities of the listening room.

**Volume Control Characteristics** — Before studying volume control characteristics, the loudness sensation of the human ear should be taken into consideration. Suppose we have a perfect speaker, into which we feed a constant amplitude signal which varies from a very low frequency to a very high frequency. Now assume that the speaker has a flat response at all intensity levels. The response of the ear is not flat at all intensity levels and resembles the response curves shown in Figure 17.

If the signal level was very high or loud the response of the ear would be flat from 25 cps to 1,000 cps, reaching a maximum at 4,000 cps and above 4,000 cps the sounds from the higher frequencies would be softer. If the signal input to the speaker is reduced from 120 db to a moderate 60 db, which would better typify normal listening levels, the low-frequency notes would be much softer. The frequencies increase to a maximum loudness at 4,000 cps and they then fall off rapidly. When a low signal level is fed to the speaker, the low-frequencies are barely audible. At low or soft levels, the ear only covers a narrow frequency range for intelligibility, as indicated in Figure 17. From a musical standpoint this is inadequate. When a symphony orchestra plays there are many low frequency notes which the average listener finds most pleasing. These low frequency notes cannot be heard at low signal levels, therefore, some type of compensation is nec-

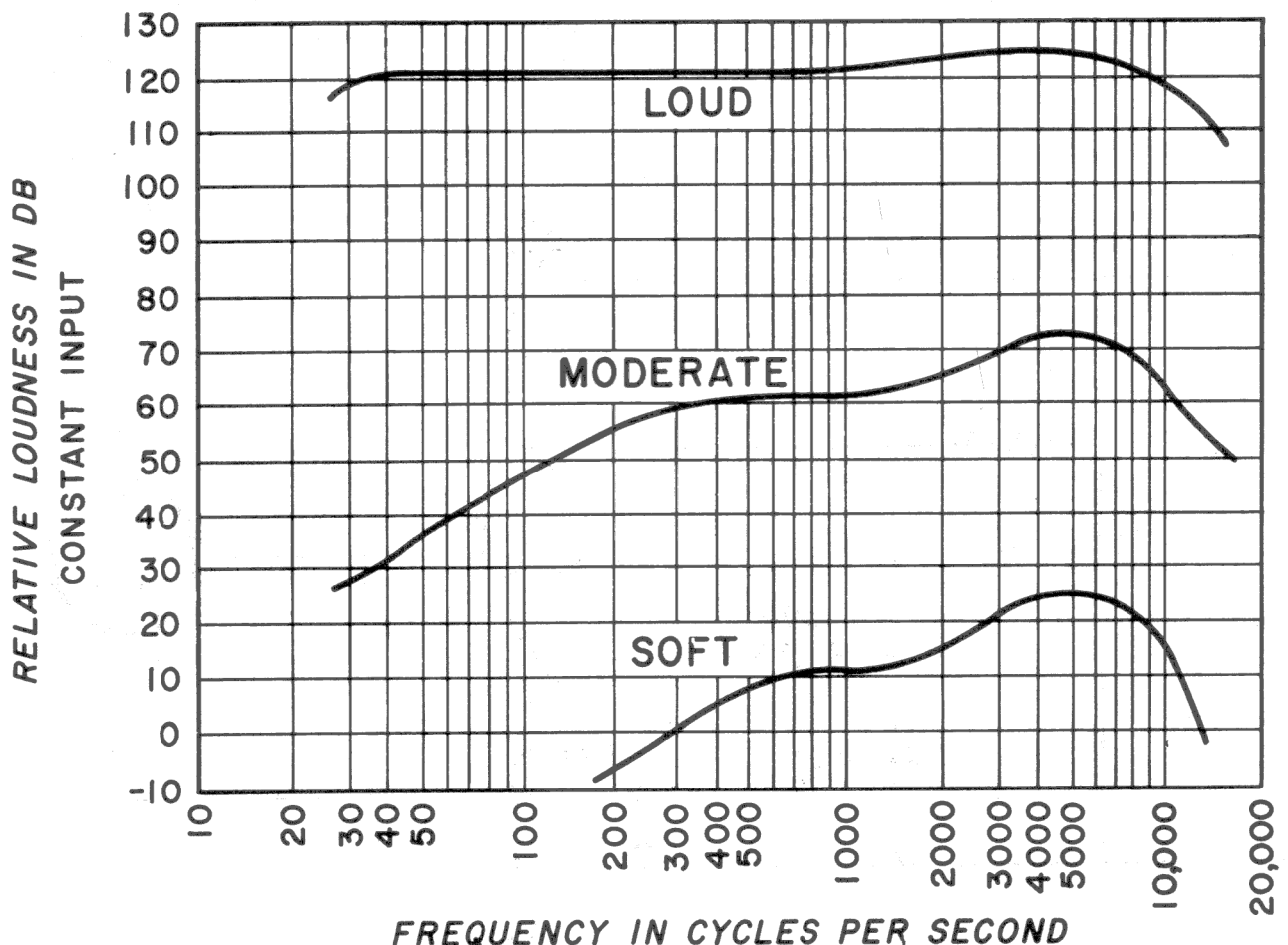


FIGURE 17

essary. This is why a *compensated volume control* is used. In many modern high-fidelity instruments the audio response is compensated at low output levels to take into account the characteristics of the ear.

The uncompensated volume control is designed so that the output of the system is flat regardless of the level which the instrument is adjusted. The bass and treble controls are adjusted by the listener. In this way the listener has complete control of the frequency range of the system at all levels. The uncompensated volume control is then considered to be non-discriminating. The loudness and volume controls which are compensated are considered frequency-discriminating devices.

**Loudness Control** — A typical loudness control circuit is illustrated in Figure 18. As the level is increased the response of the ear is flat, but as the level is lowered the response of the ear falls off considerably. Therefore, it may be desirable to make compensation at low levels. From the circuit in Figure 18 it can be seen that the loudness control is tapped at two points. At these points an r/c network is installed to ground in order to

“cut” the high frequencies in relation to the low frequencies. Thus, we are effectively compensating for the loss of low frequencies at low signal levels.

From the preceding we can conclude that at high signal levels the system is flat at all frequencies. However, at low intensity levels the system is compensated to correct for the response of the human ear.

### 34-6. TUNERS

The tuner provides the high-fidelity program material available on AM and FM radio. With two or more IF stages—improved limiter and discriminator circuits, and generally high quality vacuum tubes and circuit components, the tuner has become a standard instrument in the high-fidelity system. One of the most noticeable differences between the tuner and the AC-DC radio is the absence of a power amplifier and speaker. The output from the tuner is fed directly to the pre-amplifier or amplifier rather than to a speaker. If you examine the block diagram of a high-fidelity tuner in Figure 19, you will notice three IF stages, two limiter stages, an automatic frequency control circuit and a de-emphasis

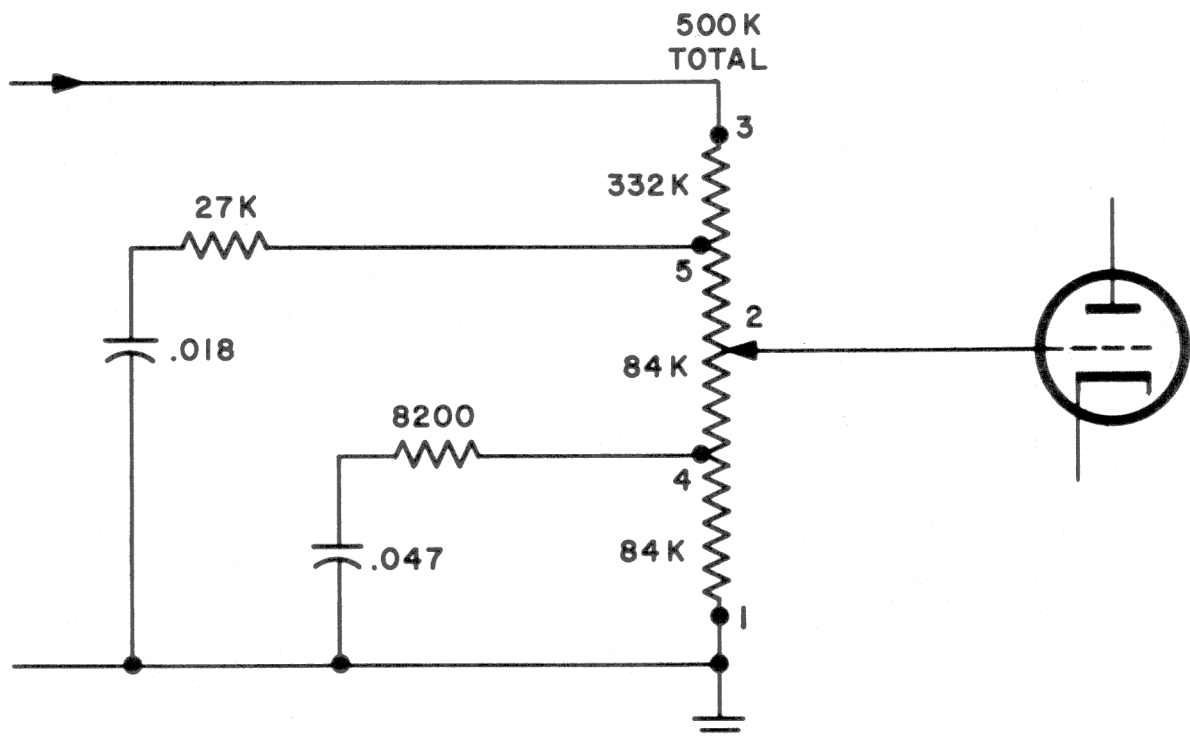


FIGURE 18



circuit at the output of the detector. These circuits help the high-fidelity tuner provide excellent sensitivity, selectivity, automatic frequency control (AFC) and detection. Therefore, they will be discussed separately in this section.

**Sensitivity** — Sensitivity is the ability of a tuner to amplify a low level signal noise-free. The degree of sensitivity is particularly important in fringe areas where good sensitivity is essential for noise-free reception. For example, a tuner rated at 5 microvolts sensitivity for 20 db of quieting provides better reception in fringe areas than one rated at 10 microvolts for 20 db of quieting, although the latter may provide ample sensitivity in metropolitan areas.

One of the most important changes that has increased tuner sensitivity is the use of RF triode grounded grid amplifiers and triode converters rather than pentodes. Due to the lack of both screen and suppressor grids, the triode produces less thermol agitation than the pentode. Thermol agitation is caused by electron bombardment on the grids which results in noise. By decreasing noise, the signal to noise ratio is increased, resulting in higher sensitivity.

**Selectivity** — The term selectivity in reference to tuners is the ability to select the desired signal while rejecting all others. The degree of selectivity must be precise, for an

over-selective tuner will be difficult to adjust to the desired station and the wide frequency response provided by FM will suffer. Whereas, a tuner with a low degree of selectivity will have adjacent channel interference caused by one station's frequency over-riding another. Tuners with excellent selectivity have low resistance high-Q coils which provide accurate frequency selections with the value of  $Q$  closely calculated to prevent the tuner from being over-selective. The number of IF stages in a tuner is another important consideration when determining the degree of selectivity. Generally, the greater the number of IF stages the better the selectivity.

**Automatic Frequency Control** — After the desired station has been selected, the problem of the oscillator drifting off the selected frequency often occurs. This frequency drift is caused by physical expansion of components due to temperature changes.

There are two common ways of compensating for this frequency drift. One method employs the use of a temperature compensated capacitor in the plate circuit of the local oscillator. As temperature changes, oscillators have a tendency to drift off frequency which results in instability. The capacitor automatically compensates for these variations in temperature and permits the oscillator to remain on frequency.

A more effective way to control this drift-

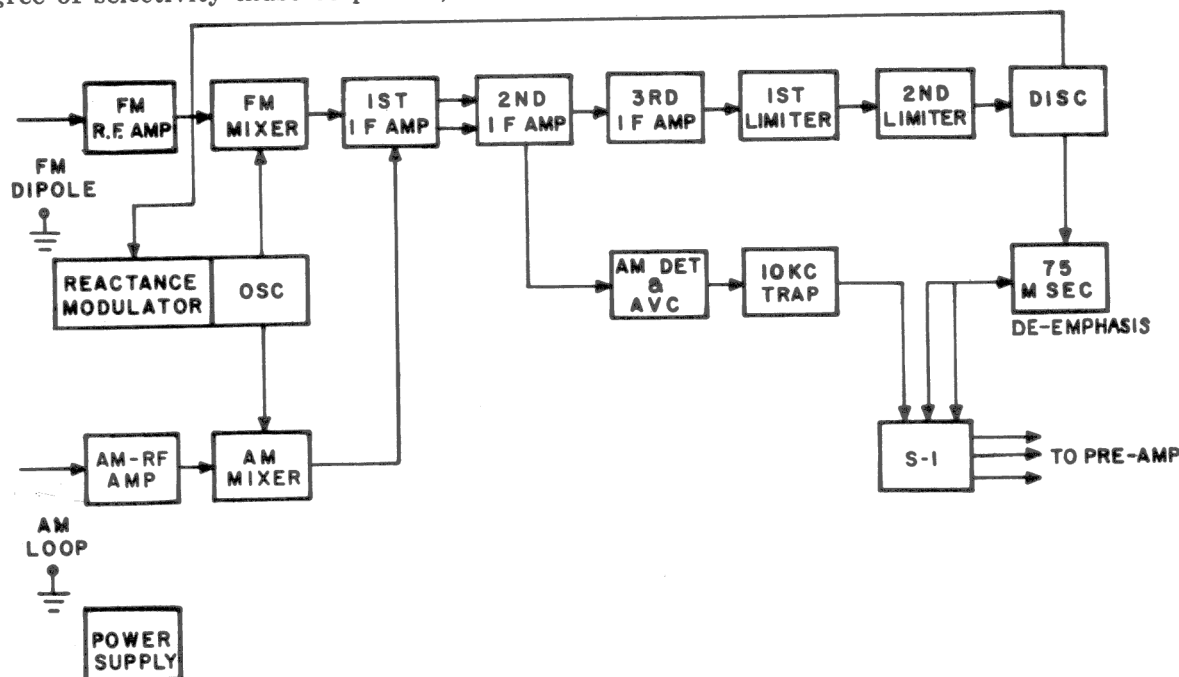


FIGURE 19

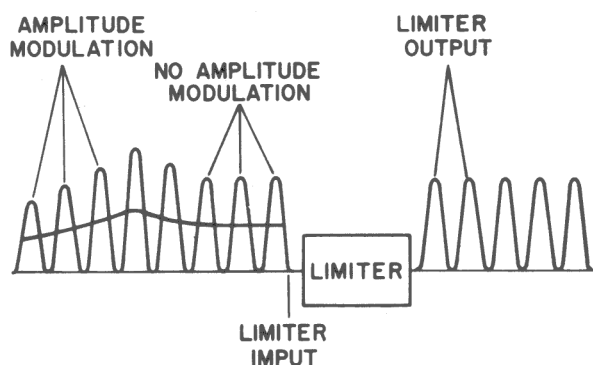


FIGURE 20

ing is to use Automatic Frequency Control (AFC). A block diagram of a tuner using AFC is shown in Figure 19. A DC voltage is taken from the output of the detector or discriminator and is fed back to a reactance tube, neon bulb or similar circuit which will change the input capacitance of the oscillator as the voltage fed into the reactance circuit varies. By changing the input capacitance, the oscillator remains on frequency regardless of the ambient temperature.

The signal present at the input of the limiter is shown in Figure 20. Note the amplitude variations at this point. Sharp cut-off pentodes having a fairly high  $G_m$  are generally selected as limiters. The limiting action is achieved through the use of grid leak biasing and low value of screen and plate voltages. Grid bias limiting is obtained by using a grid bias resistor and capacitor and low or zero voltage on the cathode. The operation of a limiter is similar to that of a grid leak detector. The average value of grid bias together with low screen voltage (approximately 50V), determine the point where plate current cut-off occurs. Any signal input voltage with excessive amplitude will cause the average grid bias to increase and, therefore, tend to hold the output voltage constant. A minimum of two volts peak is required at the limiter grid to provide sufficient amplitude limiting. In most cases the peak voltage on the grid will be on the order of 10 to 20 volts to obtain best results under poor receiving conditions. The resultant limiter output is shown in Figure 20.

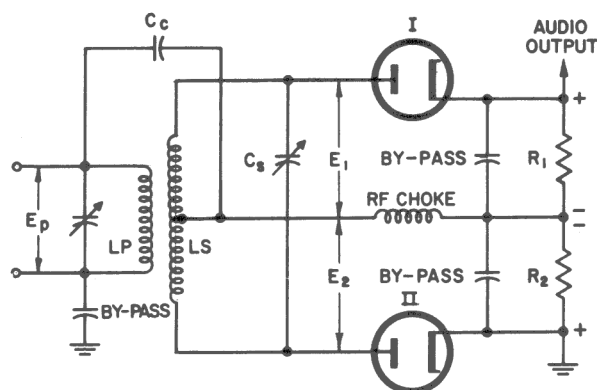


FIGURE 21

**FM Detectors** — The most common types of detectors used in FM tuners are the ratio detector and discriminator. The ratio detector responds only to changes in the ratio of the voltages across the two diode tubes and is, therefore, insensitive to changes in the differences in the voltages due to possible amplitude modulation of the RF carrier. The ratio detector has some self-contained or inherent amplitude-limiting action and may be used without any IF limiter stages as a matter of economy. In general applications, the ratio detector does not limit sufficiently, therefore, some type tuners use at least one separate stage of limiting. The limiter(s) clip off the amplitude variations resulting in a constant output level. When a ratio detector is used in conjunction with a limiter, the signal-to-noise ratio is higher with increased sensitivity as compared with the ratio detector working by itself.

Discriminators are often used as detectors in FM tuners. They are sensitive to amplitude variations, noise, static and etc., therefore, it is essential that they be preceded by one or more limiter stages. A typical discriminator circuit is illustrated in Figure 21.

Impressed across diode #1 and  $R_1$  are the sum of the primary voltage and one half of the secondary voltage due to the center tap of the secondary winding as illustrated in Figure 21. The polarity of the diode is such that the output voltage is always positive across  $R_1$ . The voltage across diode #2 and  $R_2$  is the sum of the primary voltage and the other half of the secondary voltage. The voltage  $E_1$  is positive with respect to  $E_2$ ,

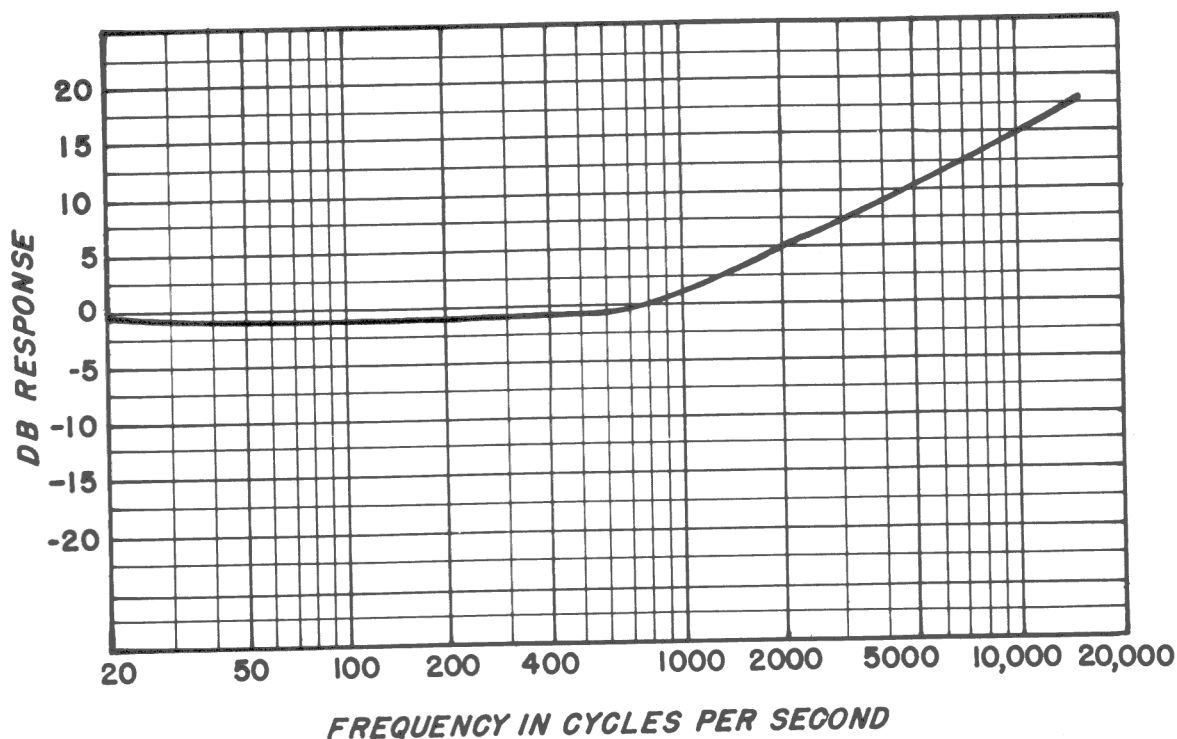


FIGURE 22

therefore, the voltage across  $R_2$  is opposite to that of  $R_1$ . The output voltage is thus the difference between the voltage appearing across  $R_1$  and  $R_2$ .

**FM Pre-emphasis and De-emphasis**—From the previous discussion of pre-emphasis during the recording operation it can be concluded that the primary reason for its use is to boost the gain of the high-frequencies during the recording process to over-ride the surface noise on the record. In FM modulation the signal is pre-emphasized before it is modulated at the transmitter to produce the same effect. The high-frequencies have a pre-emphasis as indicated by the curve shown in Figure 22. To obtain a flat response at the output of the tuner, a de-emphasis network must be utilized. For example, consider the de-emphasis circuit shown in Figure 23 and the de-emphasis curve in Figure 24. The value of the series resistance  $R_1$  is 39K. The value of  $C_1$  can be determined with the following information:

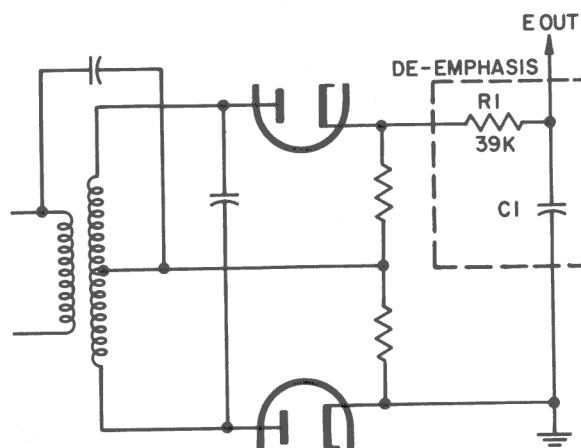


FIGURE 23

$$R_1 = 39K$$

$$17 \text{ DB at } 15KC = \text{TIME CONSTANT } 75 \text{ MICROSECONDS}$$

$$C_1 = 75 \text{ Microseconds divided by } R_1$$

$$C_1 = (75 \times 10^{-6}) \div 39 \times 10^3 = .0019 \text{ MFDS}$$

$$C_1 = .002 \text{ MFDS (nominal)}$$

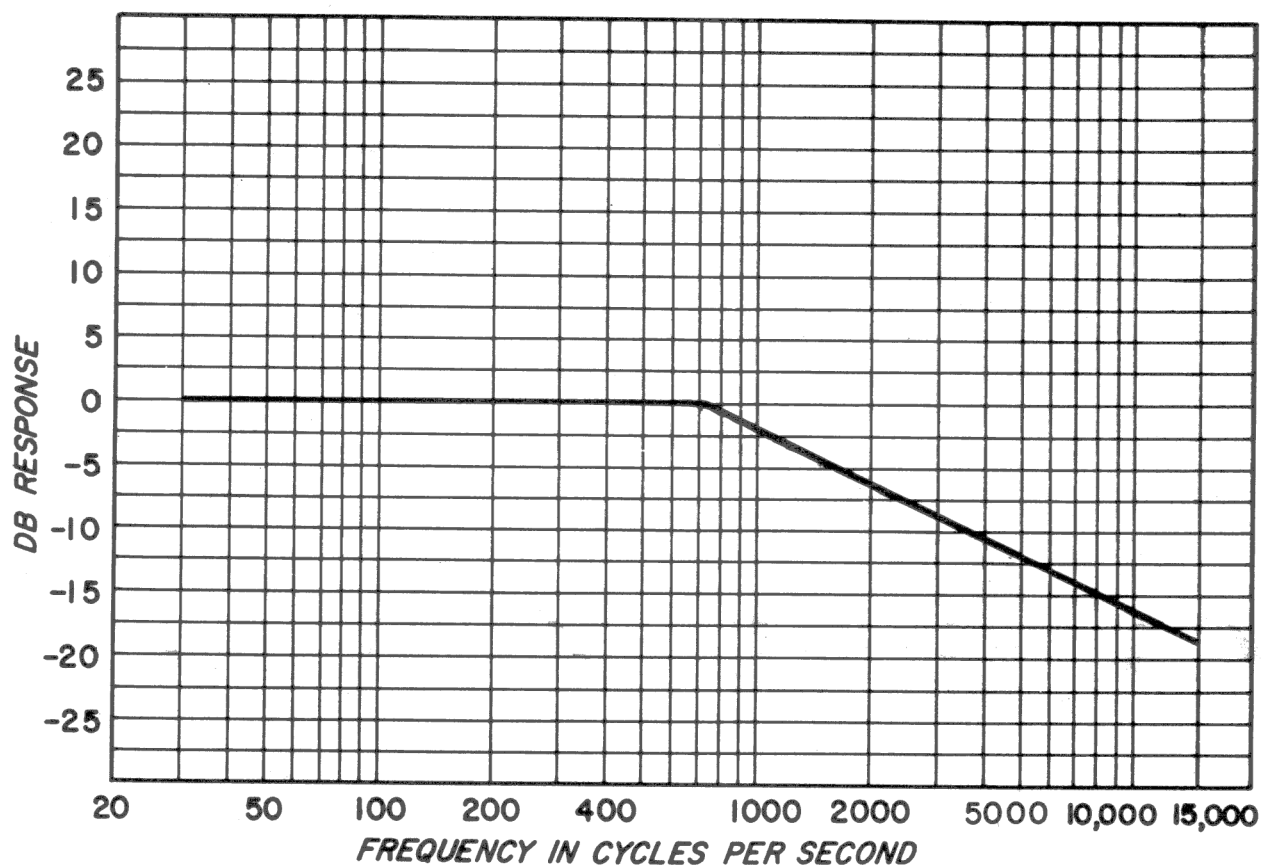


FIGURE 24

When resistor  $R_1$  is 39K and capacitor  $C_1$  is .002 MFD's the high frequencies will be attenuated and the response curve will be similar to the one shown in Figure 24. At the same time the IF carrier frequency is filtered by this de-emphasis network.

### 34-7. POWER AMPLIFIERS

Power amplifiers in high-fidelity systems are designed to provide sufficient power to drive one or more speakers. They generally make use of beam power pentodes connected in push-pull. The output stage can be operated at various classes of operation.

**Class "A"** — The Class "A" amplifier is one in which the grid is biased such that the negative signal swing does not exceed cut-off and the positive swing does not exceed the negative DC grid bias. Thus, a relatively large amount of plate current flows with no signal present at the grid. Therefore, a Class "A" amplifier has a high static plate power dissipation. This inherent characteristic results in poor efficiency due to its limited peak power output. A Class "A" amplifier is used where medium power output and very low

distortion are required.

**Class "AB<sub>1</sub>"** — A Class "AB<sub>1</sub>" amplifier operates at the bottom curved portion or instep of the Eg-Ip curve. The average plate power dissipation is equal to a Class "A" amplifier, however, the static plate power dissipation is less than that of a Class "A" amplifier. Therefore, this amplifier has greater peak power capabilities. The positive signal swing does not exceed the DC grid bias voltage, however, the negative swing can cause plate current cut-off. At no time during the positive signal swing does the grid draw current. Audio power amplifiers operating Class "AB<sub>1</sub>" have proven very satisfactory because of their medium to high peak power output characteristic and comparatively low harmonic distortion.

**Class "AB<sub>2</sub>"** — A Class "AB<sub>2</sub>" amplifier differs from a Class "AB<sub>1</sub>" in that the positive peak signal voltage is greater than the dc bias voltage, therefore, grid current is drawn and power is consumed in the grid circuit. These operating characteristics permit a Class "AB<sub>2</sub>" amplifier to obtain high peak

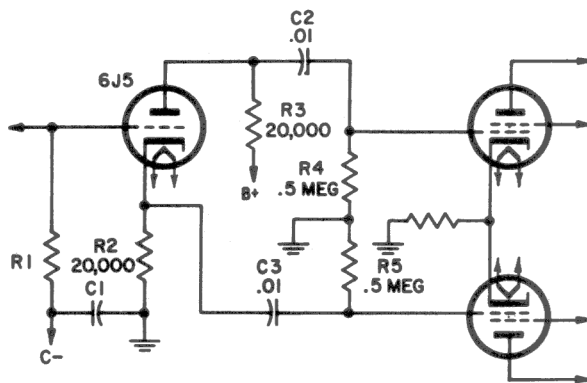


FIGURE 25

power output, although it produces somewhat higher harmonic distortion. The average plate dissipation is the same as the previous classes of operation.

Most power output stages in high-fidelity amplifiers are operated Class "AB<sub>1</sub>" because they provide ample power and low distortion. Classes "B" and "C" amplifiers are very seldom used in high-fidelity amplifiers because of their inherent harmonic distortion.

**Phase Inverters** — The purpose of the phase inverter is to provide two voltages equal in amplitude and 180 degrees out of phase, which can be utilized to drive a push-pull output stage. Figure 25 illustrates a typical phase inverter circuit. A triode is used having the same value plate and cathode resistance. When the positive half cycle of the AC signal is applied to the grid of the triode, the current in the resistors ( $R_2$  and  $R_3$ ) increases. This increase in current causes an increase in voltage drop across the resistors. As a result the plate voltage becomes more negative with respect to ground and the voltage at the cathode becomes positive in respect to ground. With the values of both cathode and plate resistors equal, the peak negative voltage at the plate is equal to the peak positive voltage at the cathode. Since these two voltages are 180 degrees out of phase, they may be used to drive a push-pull amplifier.

**Methods of Coupling** — There are various ways of coupling one amplifier stage to another, such as, transformer, resistance and direct coupling. The recent trend has been resistance and direct coupling, which have comparable performance to other methods

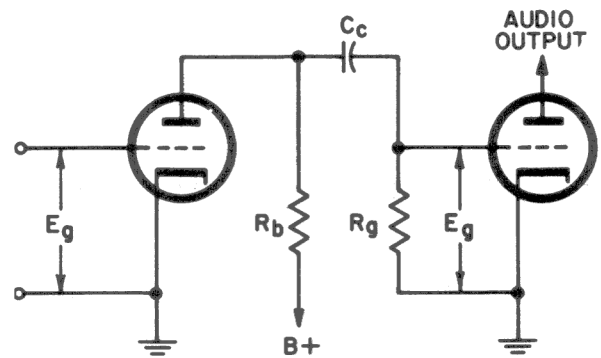


FIGURE 26

but are much less expensive. Resistance coupling involves a network of resistors and capacitors between the plate circuit of the previous stage and the grid circuit of the following tube. A typical resistance coupled amplifier is shown in Figure 26. The function of the coupling capacitor is to pass the audio signal on to the next stage and at the same time isolate the DC plate voltage from the grid of the next stage. The reactance of the capacitor is not the same at all frequencies and, therefore, determines the low frequency response at the low end of the spectrum.

The direct-coupled circuit in Figure 27 is essentially a resistance coupled circuit without the capacitors, although capacitors may be shunted across the coupling resistors to increase high frequency response. A DC plate voltage is applied to the grid of the following stage. The plate voltage of the following stage is then raised higher so that a proper relationship can be maintained between the grid and plate. The plate and grid

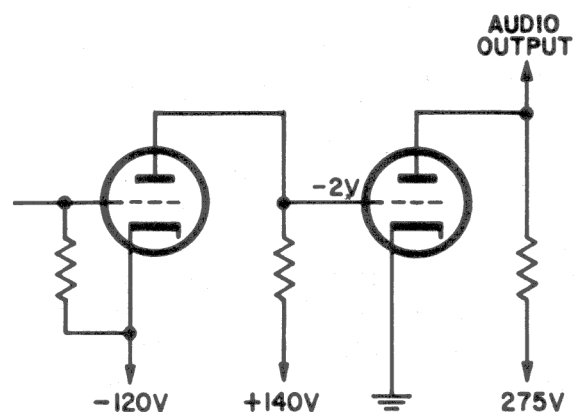


FIGURE 27

voltages are then increased proportionally from the first input amplifier stage to the final. This type of coupling requires a higher than normal voltage supply. The direct-coupled amplifier has excellent low frequency response and minimum phase distortion.

**Harmonic Distortion** — Harmonic distortion, by definition, is the production of frequencies which are a direct multiple of the fundamental frequency. For example, if a 1KC sinewave were fed into a non-linear amplifier the output may contain the second harmonic (2KC), the third harmonic (3KC), as well as the fundamental frequency. The only frequency present at the input was 1KC, therefore, the production of both the second and third harmonics is a result of non-linearity.

All musical instruments produce complex sounds containing many harmonics or overtones. Since all sounds have certain relationships between the fundamental and harmonics, it is this relationship that gives the instrument its particular sound quality. If certain harmonics are overly stressed while others are unduly suppressed, the characteristic tonal quality of a musical tone can be drastically changed. For example, if you were listening to a particular instrument being played through a non-linear amplifier, the tonal quality of the instrument could be changed to such an extent that you would not recognize the instrument.

The ability of the ear to distinguish harmonic distortion depends on the dynamic range of the reproduced sounds and the volume level. Thus, harmonic distortion would be less noticeable at wide dynamic ranges, than at a limited frequency range of the same volume..

**Phase Distortion** — Phase distortion is a less frequent trouble in amplifiers, because although it may be present it is very difficult to identify even by the most critical listeners. This type of distortion is caused by a delay that some components have on certain frequencies, while allowing other frequencies to pass undelayed. For instance, if two frequencies (100 cps, 1,000 cps) were simultaneously fed into an amplifier, and a coupling capacitor delayed the 100 cps signal by  $\frac{1}{4}$  cycle, the low frequency signal would be  $90^\circ$  out of phase with the original input signal and cause phase distortion. Similarly, a signal delayed  $\frac{1}{2}$  cycle would be  $180^\circ$  out of phase and a one cycle delay  $360^\circ$  out of phase. The

more the delay the greater the phase distortion. The primary cause of phase distortion is in the coupling components whether they be capacitors, inductors, resistors, or transformers depending on how they are used in the circuit.

**Intermodulation Distortion**—If a record containing music by a symphony orchestra were being played back through an amplifier the resultant waveform would be very complex. A serious type of distortion which can occur under these conditions is known as intermodulation. This occurs when two or more frequencies are fed into a non-linear amplifier which develops sum and difference frequencies in the output. For example, if we were to apply two fundamental frequencies  $F_1$  and  $F_2$ , to a non-linear amplifier, the output would contain  $F_1$ ,  $F_2$ ,  $F_1 - F_2$ ,  $F_1 + F_2$ ,  $2F_1$ ,  $2F_2$ ,  $2F_1 - 2F_2$ ,  $2F_1 + 2F_2$ , etc. This is a very complex waveform. The resultant sum and difference frequencies have no relation to the fundamental frequencies present at the input of the amplifier. Therefore, their presence in the audio system is undesirable and the result is intermodulation distortion which is the most displeasing of all types to the human ear. The high-fidelity manufacturers rate intermodulation distortion as being a certain percent at the rated output, with some of the better amplifiers rated from .5% to 2% intermodulation distortion.

**Power Requirements** — Under normal listening conditions in the average room, the power output of a high-fidelity system is on the order of one watt or less. But there are instantaneous peaks which drive the power output to several times this value. Therefore, the question arises, how much power output is required and why? If an amplifier can not faithfully reproduce these high peaks without clipping or overloading, then distortion will result. An amplifier should be able to develop peaks of power *seven* times the normal listening level. It is, therefore, necessary that any high-fidelity amplifier be capable of ten to fifteen watts output. There are other reasons why it is advantageous to have a larger output than required. For example, the graph in Figure 28 shows intermodulation distortion versus power output. You will note that there are two curves, one representing a 10-watt amplifier, the other a 20-watt amplifier. The amplifiers are identical except

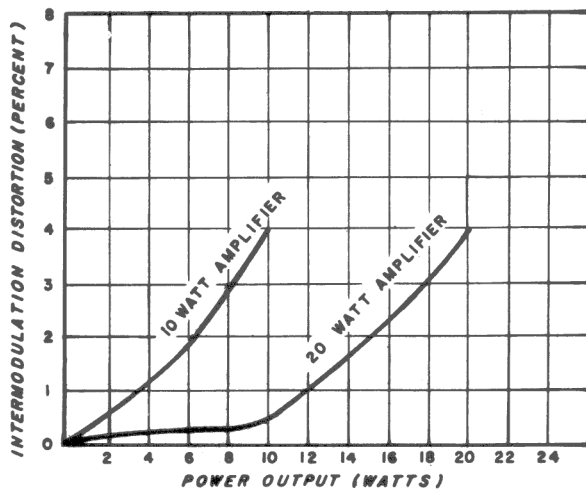


FIGURE 28

for their power output. It can be seen that the 20-watt amplifier has  $\frac{1}{2}\%$  intermodulation distortion at ten watts as compared to the 10-watt amplifier having 4% distortion at maximum output. This brings out the very important point that as an amplifier's power

is increased to its full output rating, all types of distortion, noise, etc., are maximum. Further, an amplifier with a rated output of 40 watts working at normal listening levels will have but a fraction of the distortion as the amplifiers previously mentioned. The superiority lies primarily in the fact that distortion of all types at normal listening levels is lower in the larger amplifier.

**Frequency Response and Feedback**—One of the most important factors in amplifier design is frequency response. The frequency range of any amplifier should at least cover the frequencies which the human ear is capable of hearing. The average person's hearing range is from 30 cps - 15,000 cps. Amplifiers that will faithfully reproduce frequencies from 30 cps to 15 kc are adequate for high-fidelity use. If an amplifier cannot reproduce this audio spectrum then limited frequency reproduction will result due to cutting off at the extreme low or high frequencies. Figure 29 shows a response curve of an amplifier that has limited frequency reproduction at both the high and low fre-

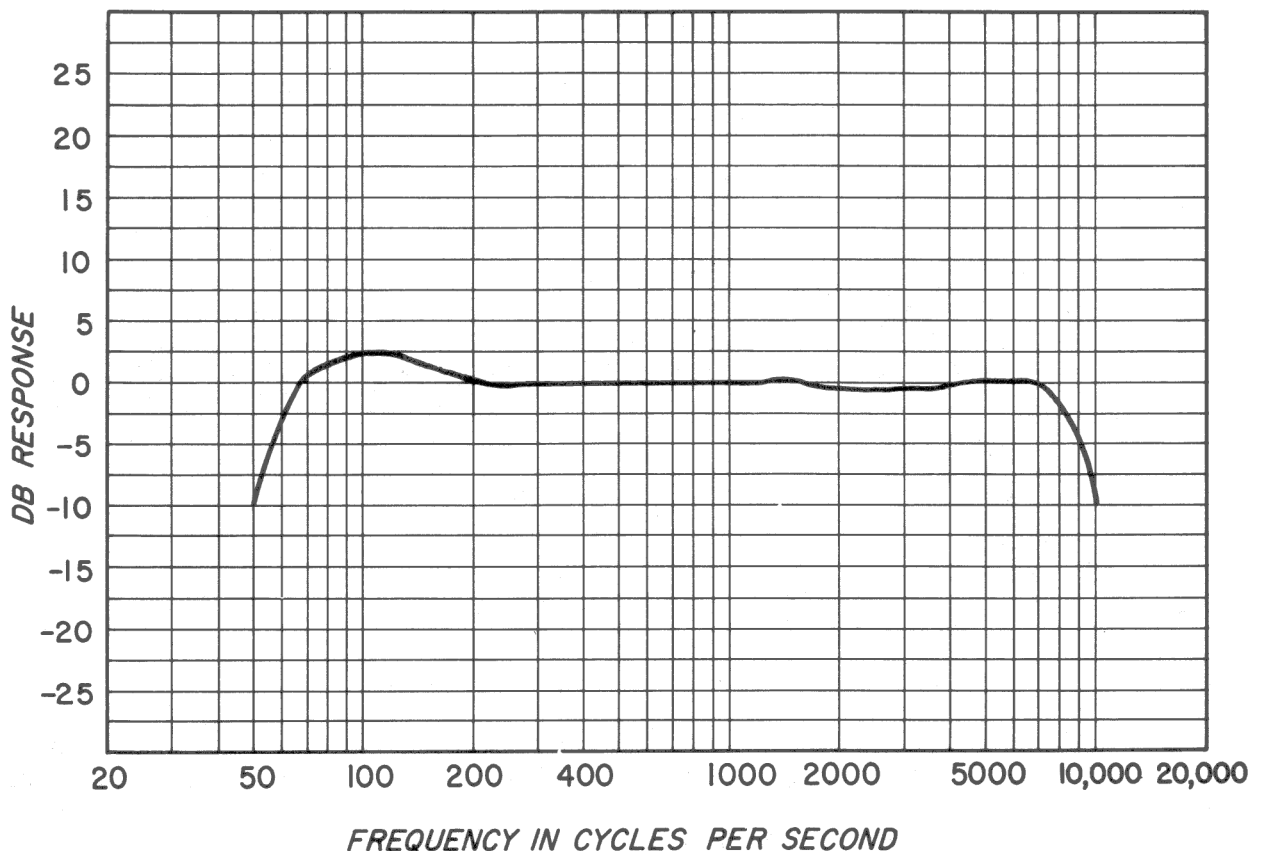


FIGURE 29

quencies. If the response is suspected of falling off at the low or high frequency ends, a good check would be to play a test record, such as, "Adventures in High Fidelity" (side #2) recorded by RCA Victor. If such a record is not available, the response can be checked, using a square-wave or sine-wave generator and oscilloscope. The generator range should be from 20 cps to 20,000 cps with 1% distortion or less.

The frequency response of an amplifier is improved with the addition of feedback. There are two types of feedbacks used in hi-fi amplifiers: positive (regenerative); negative (degenerative) feedback. A typical negative feedback loop is illustrated in Figure 30. Some of the advantages of negative feedback are as follows:

1. The frequency response curve is improved.
2. Greater stability—because the characteristics of the circuit will remain constant for wide changes in tube characteristics and voltages,
3. Reduction of harmonic, intermodulation and phase distortion.
4. Reduction of noise.
5. Decrease the ratio of impedance between the output stages and the voice coil of the speaker.

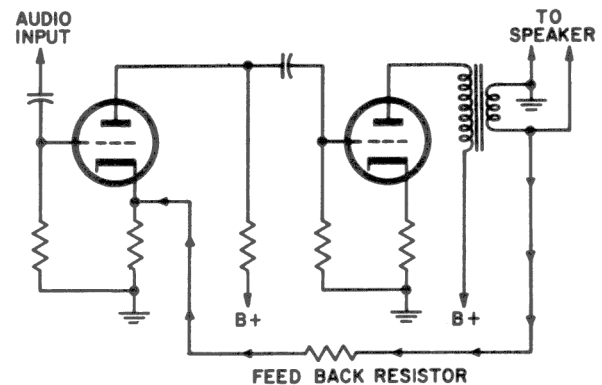


FIGURE 30

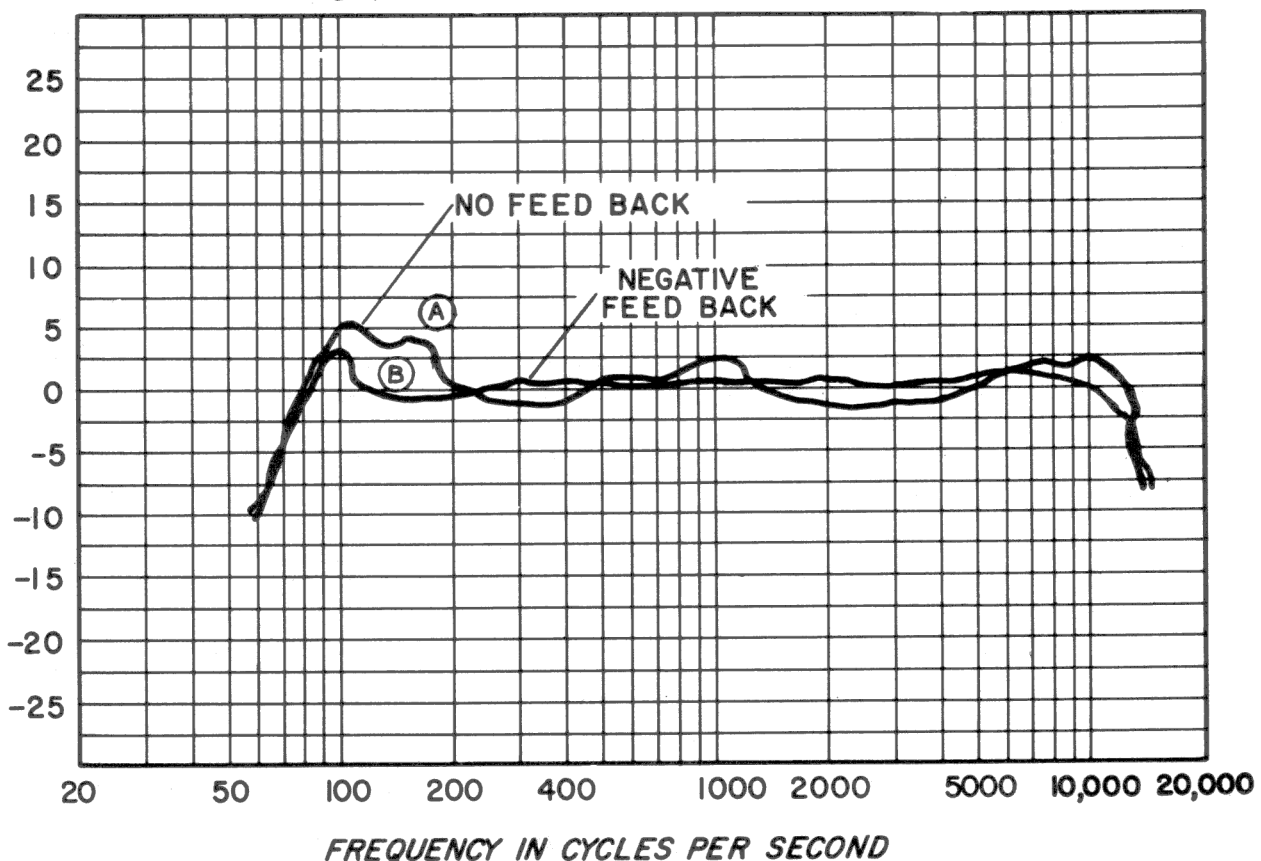


FIGURE 31



### 34-8. HIGH-FIDELITY SPEAKERS AND CROSSOVER NETWORKS

The speaker is probably the weakest link of the high-fidelity system components. At the present time, it is almost impossible for an ordinary single-cone speaker to reproduce the entire frequency range required, therefore, the speaker manufacturers have developed various speaker systems for high-fidelity use.

The tweeter, or high-frequency speaker is frequently used to reproduce frequencies from 2,000 to 15,000 cps. Such a tweeter is illustrated in Figure 32. The frequency range of a specific tweeter is only determined after extensive experiments. In general, the lower frequency limit or cut-off point of a tweeter is adjusted electrically or mechanically, at the frequency where the low-frequency or mono-range speaker stops reproducing. For purposes of illustration, we will use 2,000 cps as the cut-off point. Curve A in Figure 33 shows the frequency range of the tweeter. Curve B shows the curve of a

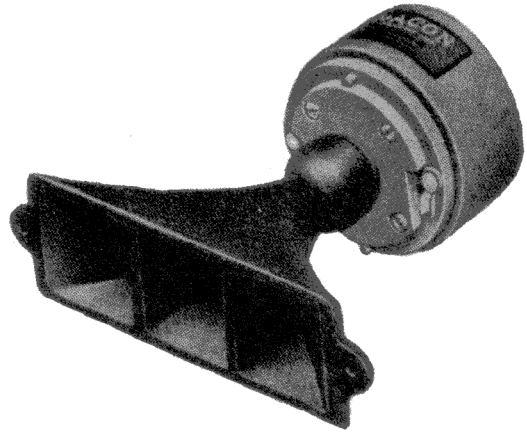


FIGURE 32

limited frequency speaker or mono-range speaker. Curve C illustrates the response of the combination. From this graph, the improvement in frequency range becomes quite evident.

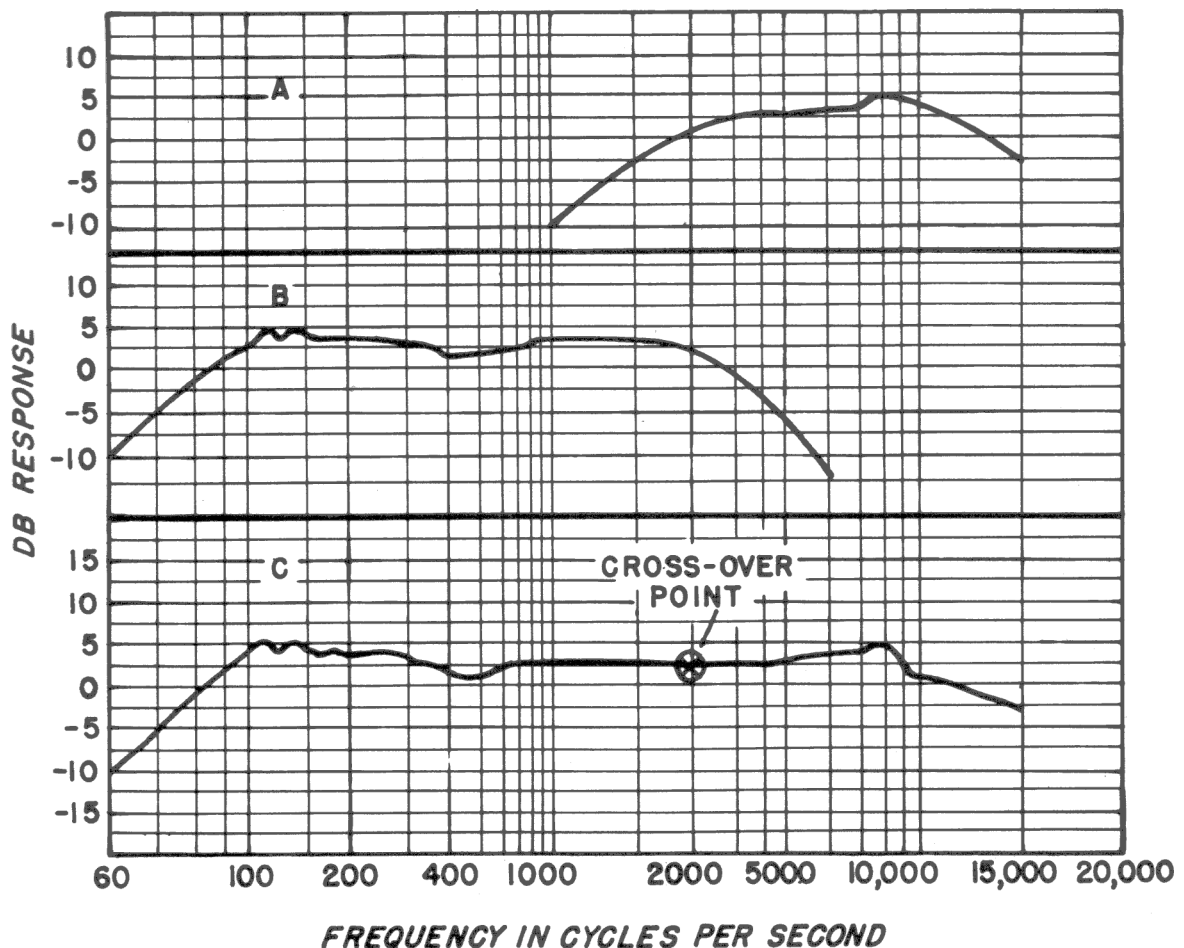


FIGURE 33

Improved frequency response is not the only advantage when using two speakers. The dispersion angles of a mono-range speaker are illustrated at various frequencies in Figure 34. Note that the dispersion angle decreases as the frequency is increased. Thus, the high-frequencies are confined to a comparatively narrow angle. Therefore, part of the listening room area will be to one side or the other of the dispersion angle and listeners in this area will hear only a limited frequency range. The use of two speakers helps to overcome this problem of limited high frequency dispersion, thus increasing the listening area.

The combination of two speakers has improved the frequency range and dispersion angle but at the same time has created a problem. We previously mentioned the cut-off or cross-over point. Referring to Figure 33, it can be seen that the frequency range of the mono-range speaker (or low-frequency unit) extended above 2,000 cps, the frequency we selected for the cut-off (cross-over) point. The tweeter's frequency range also extended below 2,000 cps. In other words, an overlap of frequencies from the two speakers exists. If not corrected, there will be an increase in amplitude at the cross-over point. This means that the frequency response of the speaker system will no longer be flat. It is necessary to prevent the low-range speaker from producing frequencies above 2,000 cps and the tweeter from reproducing frequencies below 2,000 cps. The device used to prevent one frequency from overlapping another is known as *cross-over network*. The cross-over network determines the

frequency at which the cross-over of reproduction from one speaker to another takes place. A typical cross-over network is shown in Figure 35. The choice of a cross-over point or points is not a haphazard decision but a result of careful analysis of the characteristics of the speakers involved.

The first consideration is the point where the low-frequency speaker becomes non-linear. The second consideration is the dispersion angle of the low-frequency speaker and at what frequency does this angle coincide with the dispersion angle of the high-frequency tweeter. A further discussion on cross-over networks will be covered later in this section.

**The Woofer**—There are distinct advantages of using more than one speaker to reproduce the audible spectrum and we have regarded the mono-range speaker as the unit which reproduces the low and midrange frequencies. However, this is not the case in all high-fidelity speaker systems. A speaker which is designed to reproduce a limited range of frequencies in the lower register is called a *woofer*. The size of a woofer's cone varies from 12" to 18", depending on the specific application. Generally the larger the cone, the lower the frequencies which the speaker can reproduce.

**Coaxial Speakers** — We have discussed three separate speakers, one reproducing the lows, one the middle range frequencies and the high-frequency tweeter which reproduces the high-frequencies. It is possible to take two different speakers and mechanically com-

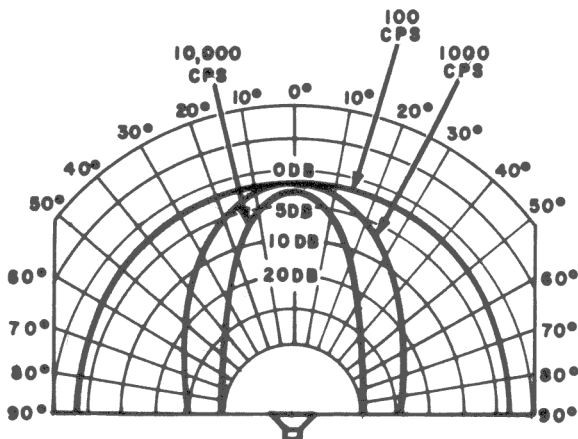


FIGURE 34

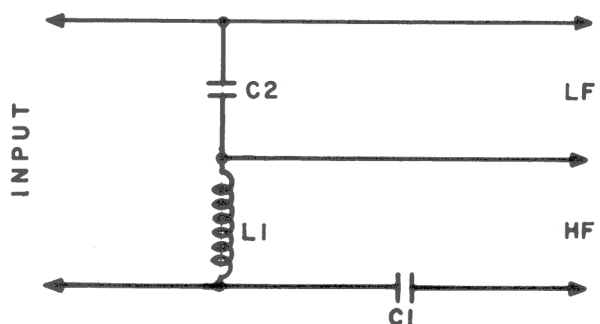


FIGURE 35

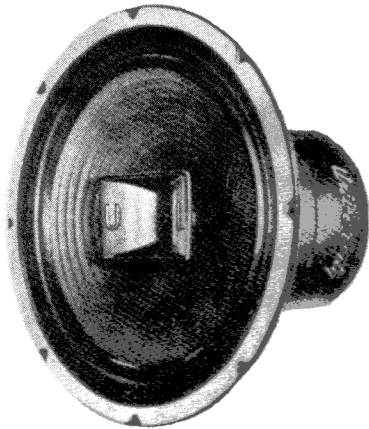


FIGURE 36

bine them into one unit. The unit is known as the *coaxial speaker*. A typical coaxial speaker is shown in Figure 36. Note that the high-frequency tweeter is mounted directly in front of the large speaker cone. The large cone reproduces the low and mid-range frequencies while the tweeter reproduces the high end of the frequency spectrum. The coaxial speaker is used extensively in high-fidelity systems and provides superior sound reproduction as compared with one single cone speaker. The curve shown in Figure 37 illustrates the response curves of two speakers: "A" representing the single cone speaker; "B" showing the response of the coaxial type speaker. The coaxial speaker uses a cross-over point near 2,000 cps.

**Triaxial Speakers** — Three speaker units can be mechanically combined to form a single speaker. Such a unit containing low-frequency, mid-range, and high-frequency components is called a *triaxial speaker*. A cut-away drawing showing the assembly of a

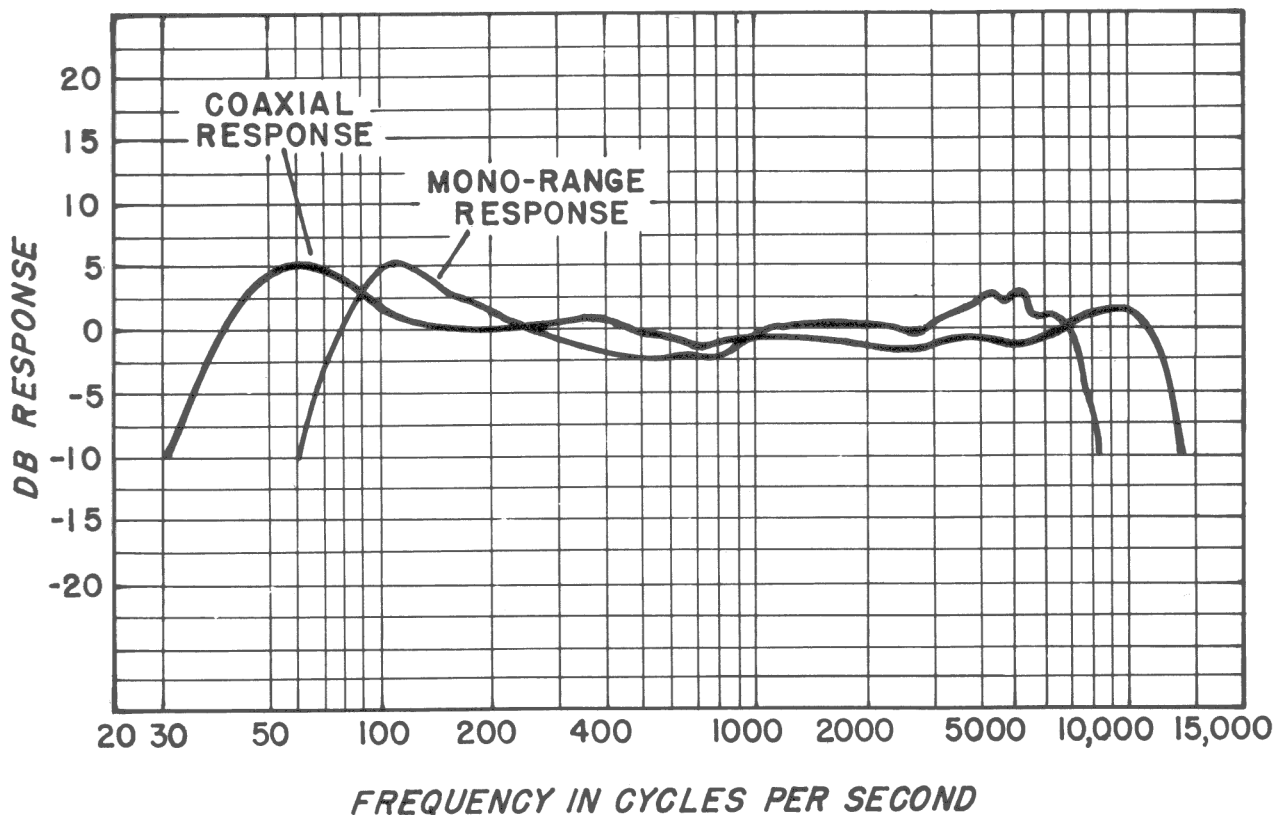


FIGURE 37

triaxial speaker is illustrated in Figure 38. The cross-over frequency for the coaxial speaker is 2,000 cps, whereas most triaxial speakers have cross-over points at 600 cps and 4,000 cps.

**Cross-Over Networks** — There are various types of cross-over networks ranging from the most simple, containing a capacitor in series with the high-frequency speaker, to the more complex networks containing two capacitors and inductors as shown in Figure 39. The circuit shown in Figure 39 provides a constant load impedance to the amplifier. This circuit contains a high pass filter in parallel with a low pass filter. The high pass filter ( $C_1$  and  $L_1$ ) allows the high-frequencies to pass and rejects the low frequencies. The low pass filter ( $C_2$  and  $L_2$ ) passes the low frequencies and rejects the high frequencies.

The cross-over network is designed specifically for each speaker system depending on the frequency range of each speaker. For example, one of RCA's speaker systems uses a 12" speaker and two 3½" speakers (tweeters) to reproduce the audio spectrum. The design of the 12" speaker is such that it is capable of reproducing the low, mid-range and high frequencies. The high frequencies are not prevented from being reproduced by the 12" speaker since it is capable of reproducing the high frequencies distortion free but at a relatively narrow dispersion angle. The cross-over network used with this speaker is illustrated in Figure 40. Capacitor  $C_1$  allows only the high frequencies to be fed to the tweeters. Wide angle high frequency dispersion is provided by the tweeters which are mounted pointing away from the center of the enclosure. The area directly in front of the enclosure is covered by the high frequency radiation from the 12" speaker.

**Types of Speaker Distortion** — There are various types of distortion created by the speaker which fall under the same title as those previously mentioned in amplifiers. Before we go any further let's define distortion as those sounds which are part of the output signal, but were not contained in the original signal. The types of distortion most commonly found in speakers are *phase*, *transient*, *amplitude* and *intermodulation*.

*Phase distortion* is the inability of the

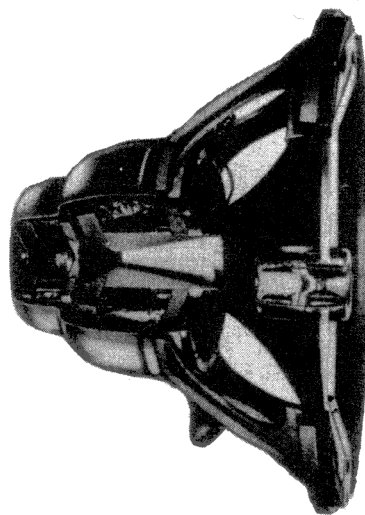


FIGURE 38

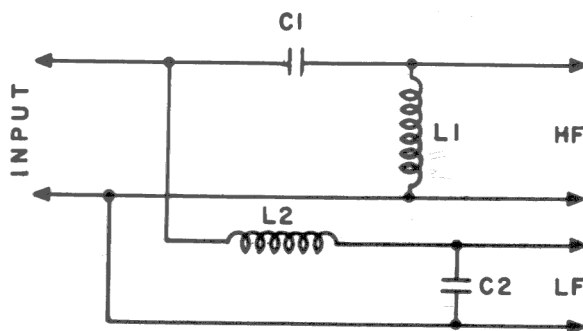


FIGURE 39

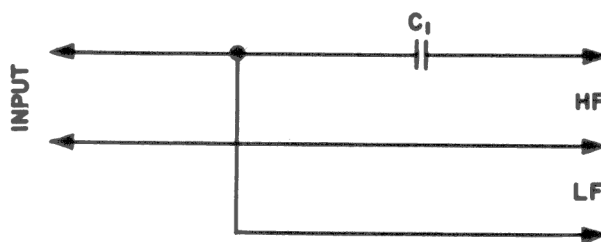


FIGURE 40

speaker to react equally to all frequencies. The total speaker impedance is the resultant inductive reactance and resistance of the voice coil. From the formula for  $X_L$  ( $2\pi fL$ ) it can be noted that as frequency increases  $X_L$

increases. Thus, if the inductive reactance ( $X_L$ ) is much greater than  $R$  at all frequencies no phase distortion will result. However, if at low frequencies  $R$  approaches  $X_L$  then a noticeable amount of phase distortion may result. Therefore, the voice coil should have a high- $Q$  factor and be non-resonant within the audio frequency spectrum.

*Transient distortion* is the inability of the speaker to smoothly follow the applied signal without producing resonant peaks or dips in the speaker response. The speaker is a mechanical device and thus it is subjected to certain amounts of inertia. If a signal were suddenly applied to a speaker, there would be a time delay until the speaker would vibrate at the applied signal. If the signal source were suddenly stopped, the speaker cone may continue vibrating for an instant because of its mechanical inertia. This gradual stopping of the vibrating cone contributes to transient distortion and is known as *acoustic hangover*. The acoustic hangover is similar to the reverberation period in that the initial impulse starts a continuing but gradually diminishing sound. The curves in Figure 41 represent transient distortion for two different speakers. (A) shows a speaker which takes a certain time to diminish in amplitude as compared to (B) which has less transient distortion. From these curves we conclude that (A) has more transient distortion because of excessive *fading time* or acoustic hangover. Transient distortion distorts the next note because the speaker cone is still in a state of vibration. This type of distortion causes a muddy reproduction of sound and is particularly noticeable during a piano solo.

Another type of speaker distortion is called "*amplitude distortion*." It is caused by non-linear response of the speaker cone. A

*non-linear response* is one in which the effect is not proportional to the cause. In other words, if we were to apply a .5 volt signal to the speaker voice coil the cone would move .1 inches; when we apply a 1 volt signal the cone should move twice as much or .2 inches. If amplitude distortion is occurring, the movement of the cone will not double even though the signal strength has been increased to twice its original value. A response curve of a speaker having amplitude distortion is illustrated in Figure 42. The signal input has higher peak swing than the cone deflection.

Amplitude distortion occurs at the lower frequencies because these frequencies cause greater displacement of the speaker cone. It is difficult to identify amplitude distortion from other types of distortion merely by listening. However, an oscilloscope can be used as an accurate check, by simply connecting the scope leads across the voice coil and gradually increasing the output of the amplifier. If the wave shape on the scope begins to flatten out as the volume is increased the speaker system exhibits amplitude distortion.

The last type of speaker distortion which we are going to discuss is *intermodulation distortion*. It is caused by a speaker which at some point in its frequency range, produces a non-linear characteristic. Undoubtedly, this is the most irritating of all types of speaker distortion to the ear.

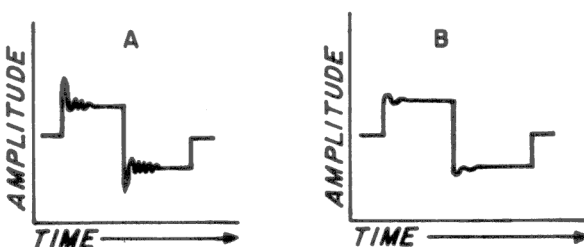


FIGURE 41

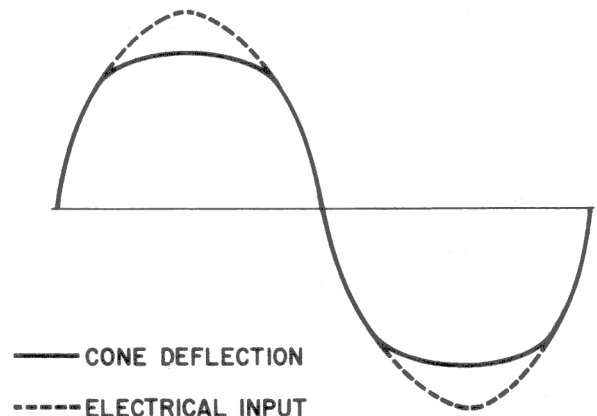


FIGURE 42

A typical example of intermodulation distortion is shown in Figure 43. As can be seen in section (A) two frequencies (100 cps and 2,500 cps) are simultaneously fed into a linear device with the resultant output containing the two original frequencies (100 cps and 2,500 cps). Section (B), Figure 43 shows the same two frequencies being fed into a non-linear device. The non-linear device has created sum and difference frequencies which were not a part of the input signal.

### 34-9. SPEAKER ENCLOSURES

No high-fidelity system is complete without some type of speaker enclosure. The enclosure is simply the housing in which the speaker(s) is mounted. Therefore, it is important that we know what effect the size, shape and location of this housing has on the sounds which you hear from an audio system.

**Bass Reflex** — The *bass reflex* is the most widely used enclosure for high-fidelity reproduction. The main reasons for the success of the bass reflex enclosure is that it is simple to construct and provides excellent frequency response. The construction of the closed cabinet is such, that a portion of the low frequency energy from the speaker is coupled to the outside air through a vent in the front of the enclosure. To obtain optimum performance, the resonant frequency of the enclosure should be lower than the resonant frequency of the speaker and the vented area should be approximately equal to that of the speaker cone. The construction of the bass reflex enclosure is shown in Figure 44. The

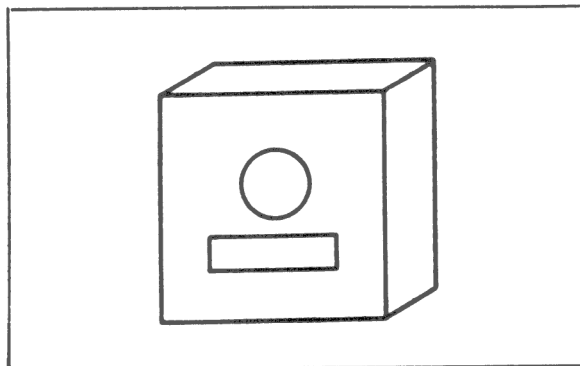


FIGURE 44

volume of air inside the enclosure offers a low impedance to low frequencies and a high impedance to high frequencies. Thus, only the low frequencies are coupled through the vented opening. The low frequencies are also radiated outward from the speaker cone. The resonant cavity inside the enclosure is such that the low frequencies from the front of the speaker cone and those from the vented opening are in phase and add together, producing an added bass response. This will in effect extend the low frequency response of the speaker. The bass reflex enclosure, when used with a speaker which has a higher resonant frequency than the enclosure produces a flatter response curve than if the situation were reversed. The flatter response makes for a more uniform sound throughout the low-frequency range where resonant peaks usually occur. The size of the enclosure is generally increased as the resonant frequency of the speaker is decreased. This will, in effect, keep the resonant frequency of the enclosure below that of the speaker and produce a smoother response.

The bass reflex enclosure has one distinct disadvantage when used with a mono-range speaker. The mid-range and high frequencies have a tendency to become phase distorted because of the phase relationship between the frontal radiated waves from the cone and the backward radiated signals from the vented opening. Acoustic padding can be used to help absorb the mid-range and high frequencies to prevent them from being radiated from the vented opening.

**Infinite Baffle** — Another type of speaker enclosure similar to the bass reflex is the *infinite baffle*. The only visual difference between the two enclosures is the absence of the vented

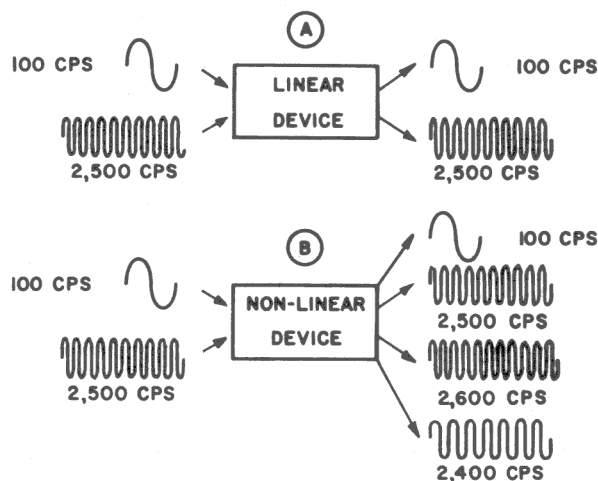


FIGURE 43

opening in the infinite baffle. The absence of a vent changes the operating characteristics of the enclosure. A typical infinite baffle is illustrated in Figure 45.

The definition of *infinite* implies that the rear waves are dispersed into a mass large enough to damp out or absorb all of these waves. In some custom installations the speaker is mounted inside a wall. This type of installation uses the air cavity of the room located directly behind the speaker to absorb the waves reflected backward. When a wall is not used the speaker is mounted inside an enclosure with the rear panel heavily padded with acoustic material. This padding absorbs the waves that are being radiated back from the speaker. The resonant frequency of the enclosure is above audibility and therefore, a low-frequency woofer should be used for maximum bass response. The overall objective is to keep the resonant frequency of the enclosure above that of the speaker and thus improve the linearity of the low-frequency response.

**Folded Horn** — The *folded horn* enclosure is popular because of its extended low-frequency response and high efficiency. The folded horn radiates the high-frequencies directly from the front of the speaker cone while the lows are coupled through the horn. To obtain the best low-frequency response, the enclosure should be placed in a corner so that the walls and floor can be used in conjunction with the horn. With the addition of the walls and floor, the system can reproduce the low-frequencies with an increase of 2 to 4 db over the standard or conventional enclosure. This extended output in the low-frequency region is an important quality because it helps to obtain a one to one ratio of low to high fre-

quencies which gives the tonal balance necessary for high-fidelity reproduction. When the enclosure is located in the corner the bass response is extended because of the additional low-frequency energy radiation provided by the walls and floor. The high-frequencies tend to beam to a point where the dispersion angle is minimized between the enclosure and the listener. In a corner location the maximum angle between the center beam of the speaker and any listening position in the room will be  $22\frac{1}{2}$  degrees. This is a minor disadvantage of the folded horn enclosure and it should be noted that if the listener is located to one side of this dispersion angle that he will not hear all of the sounds being reproduced by the system.

#### 34-10. HIGH-FIDELITY AMPLIFIER MEASUREMENTS

Frequency response, distortion and tone control and equalizer operation are the most important factors to be considered in checking high fidelity audio amplifiers. The WA-44A is extremely well suited to provide the audio signal necessary for making these checks.

Most service technicians and experimenters do not have the test set-up necessary for measuring the response of tuner, phonographs and speakers. The amplifier, on the other hand, is readily tested with high quality service instruments. Before attempting these checks make sure that the amplifier is in proper operating condition.

If measurements are taken with the speaker which will normally be used with the amplifier, the effect of the reflected speaker impedance on the total response may be noted. If it is desired to test the amplifier alone, then a dummy load consisting of a resistor of the same value as the impedance of the voice coil and of sufficient wattage to dissipate the maximum output of the amplifier, should be connected across the secondary of the output transformer.

**DB Measurements** — In order to study the relationship between the output voltage and the frequency response of an amplifier it is helpful to plot a curve showing the voltage output in decibels against frequency. The

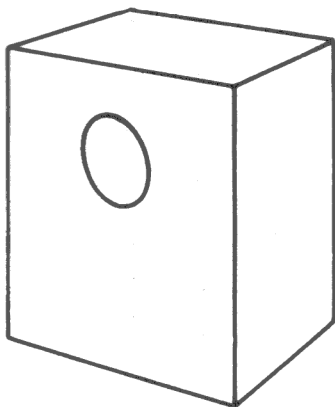


FIGURE 45

conversion table, Figure 46, offers a convenient means of changing voltage readings to decibels. Unless the same reference level and load impedance is used, the figures obtained from the chart will be relative values.

**DBM Measurements** — The graph in Figure 46 can be used to determine dbm values corresponding to rms ac-voltage values across a 600-ohm resistive load. A dbm value is defined as the number of decibels above or below a reference level of 1 milliwatt in 600 ohms at 1000 cycles. Zero dbm, therefore, would indicate a power level of 1 milliwatt; 10 dbm, 10 milliwatts; and 20 dbm, 100 milliwatts.

The graph makes possible rapid conversion of rms voltages to corresponding dbm values. Associated power levels can be read along the top of the graph. If the rms voltage is measured across a resistive load other than 600 ohms, the correction factors given

below must be added algebraically to the dbm values read from the graph in Figure 46. For resistive loads not given in the table, the following formula should be used for determining the correction factor:

$$\text{Correction Factor} = 10 \log \frac{600}{R}$$

where R is the load in ohms. If R is greater than 600 ohms, the correction factor is negative.

Because dbm are defined with respect to a 600-ohm load, power levels correspond to voltage values. DBM can be measured in terms of rms voltages across a 600-ohm resistive load. For example, 0.775 rms volt indicates 0 dbm and 7.75 rms volts indicate 20 dbm. The decibel and ear-response curves have their closest correlation at 1000 cycles.

**Frequency Response Measurements** — Three methods are generally accepted for measur-

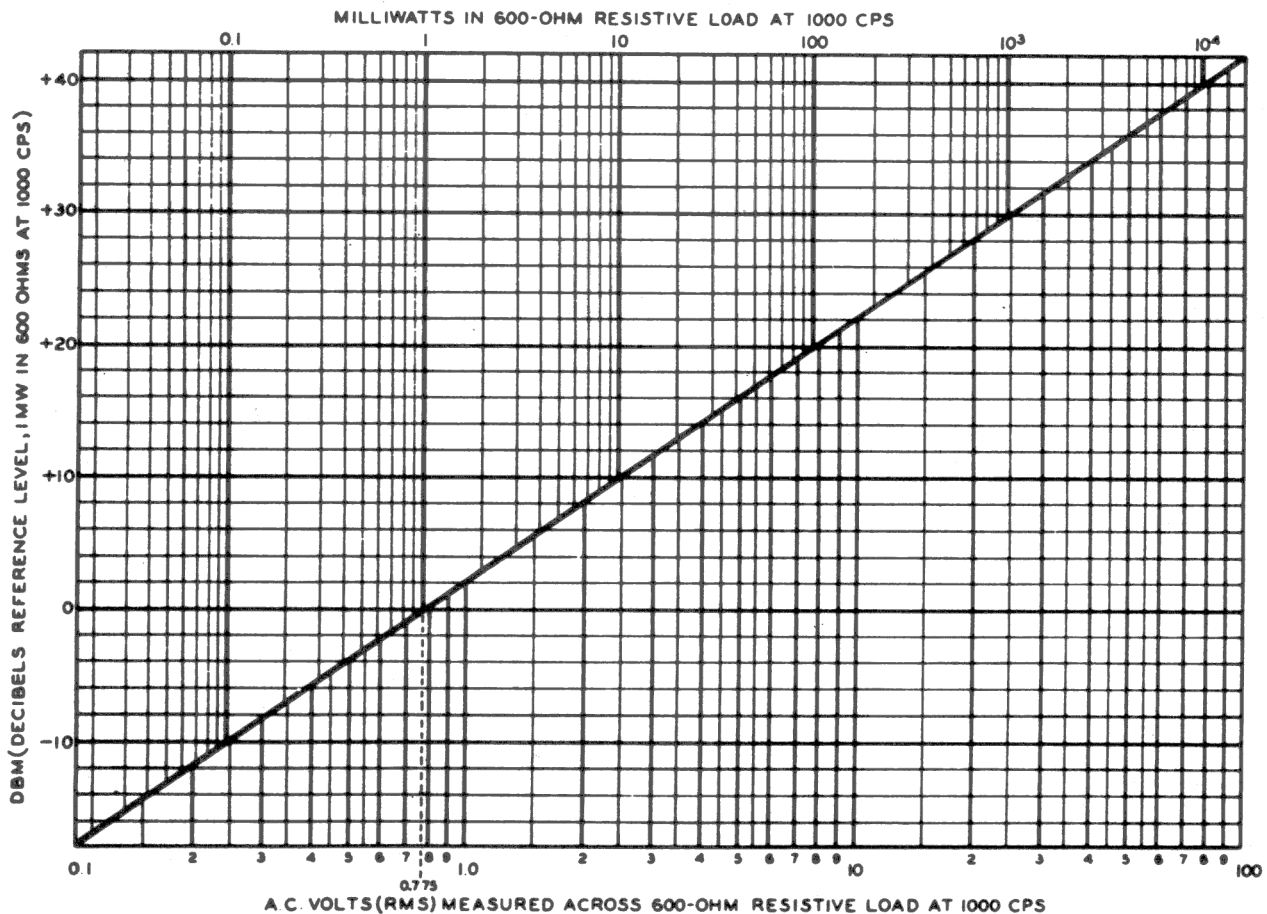


FIGURE 46



ing the frequency response of audio amplifiers. The first utilizes a sine-wave signal of known input voltage and frequency. The output voltage is measured at the voice coil and plotted (as the ordinate) against the frequency (as the abscissa). The second method is similar to the first except that an oscilloscope is used as an output indicator. This technique allows for the observation of distortion as well as output. The third method requires the use of a square-wave input signal. This technique allows for broad band measurements and eliminates the necessity of a point-by-point frequency check.

### Method 1

Connect the WA-44A to the amplifier input and adjust the output level so that maximum rated power is delivered to the voice coil.

Starting at a frequency below the estimated response of the amplifier, record the voltage reading. Increase the frequency of the audio generator in 10 cycle steps until 100 cycles is reached, 100 cycle steps until 1000 cycles is reached, and 1000 cycle steps until the limit of the amplifier response is reached. Convert the voltage readings to dbm and plot a curve of dbm versus frequency. The ideal amplifier, will have constant output voltage over the entire frequency range provided the input voltage is kept constant.

### Method 2

An oscilloscope that has been voltage calibrated may be substituted for the vtm. The measurement technique is identical with that used for the vtm. This method, however, has the added advantage of allowing a visual inspection of the output waveform. Distortion and spurious responses are readily seen on the oscilloscope.

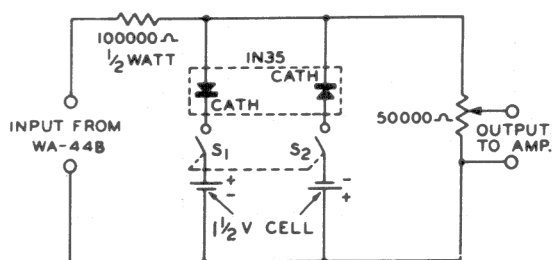
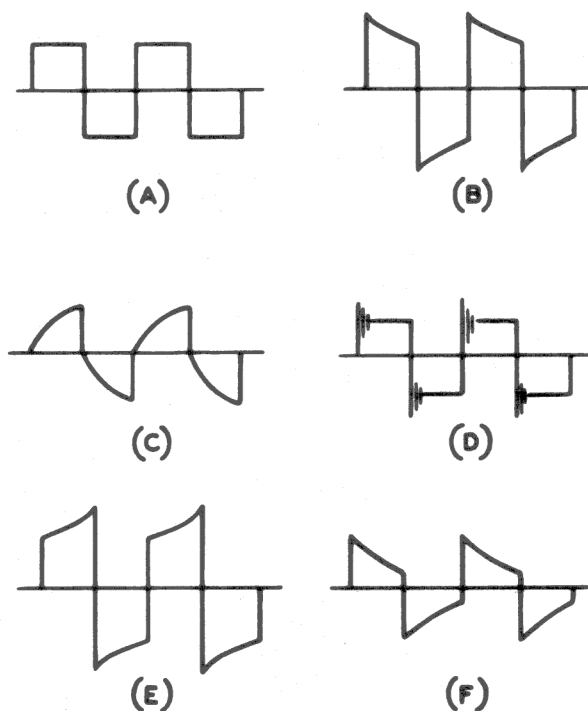


FIGURE 47

### Method 3

Square wave testing is the most rapid way of checking the response of an audio amplifier. Since a square wave is composed of many harmonics of the fundamental frequency, an amplifier that passes a square wave without distortion, has in effect responded to all of the harmonics that go to make up the square wave.

The sinusoidal output of the WA-44A may be squared by means of a simply constructed diode clipper. While the square wave produced in this manner is not as rich in harmonics as is one with a faster rise time, it should be adequate for a quick overall check of amplifier response. The clipper circuit and the equipment set up are shown in Figure 47. The traces, Figure 48, show some typical amplifier responses. Before the amplifier is checked the output of the clipper should be observed on the oscilloscope so that the input waveshape is known. The use of the square wave frequencies 20, 100, 500 and 2500 cps should be sufficient for testing amplifier response to beyond 10,000 cps.



Typical amplifier response patterns. (A) Normal square-wave (B) High-frequency boost (C) Poor high frequency response (D) High-frequency oscillations (E) Low-frequency boost (F) Poor middle-and-high-frequency response.

FIGURE 48

**Tone Controls** — If the amplifier under test incorporates tone control circuits, care must be taken to set these controls so that they have a minimum effect on the frequency response of the amplifier. In some cases these controls have a "flat" position, in other cases the point of minimum effect will have to be determined experimentally.

Set the WA-44A to a frequency in the treble range, 3000 cps is a possible starting point. Vary the bass control and notice the effect on the output voltage. Similarly, vary the treble control. If the treble control produces a large voltage change, lower the frequency of the audio generator. Alternately vary the bass and treble controls and adjust the frequency of the generator until a frequency is found at which both bass and treble controls have minimum effect on the output voltage. This frequency can be used as a reference frequency for plotting a curve of output voltage versus frequency. The effect of varying the bass control should be plotted from the lower limit of the amplifier up to the reference level, while the effect of varying the treble control should be plotted from the reference level frequency to the upper frequency limit of the amplifier.

Assume that 1000 cps has been found as the frequency at which the bass and treble control have the least effect on the output volt-

age. Set the bass control to minimum and take output voltage readings with the frequency varied from 30 to 1000 cycles. Set the bass control to maximum and repeat the output voltage readings as the frequency is varied.

The same process is followed in determining the effect of the treble control except that the output voltage reading need not be taken below 1000 cycles.

From these voltage readings four curves may be plotted. Usually they are plotted in decibels on the same base line (see Figure 46) so that comparison of the results are easily seen. These curves indicate the frequency response of the amplifier with minimum and maximum bass and treble. Intermediate positions of both controls are sometimes plotted to give an overall picture of the effect of varying these controls.

**Checking Phonograph Equalizers** — Phonograph equalizers are simple networks used to compensate for attenuation or boost in the record which is introduced during the recording process. The usual cross-over frequencies (the frequency below which the bass is attenuated) are 250, 300, 350 and 500 cps. Various record manufacturers use different cross-over frequencies, and consequently they have different recording curves. Some typical record curves are shown in Figure 49.

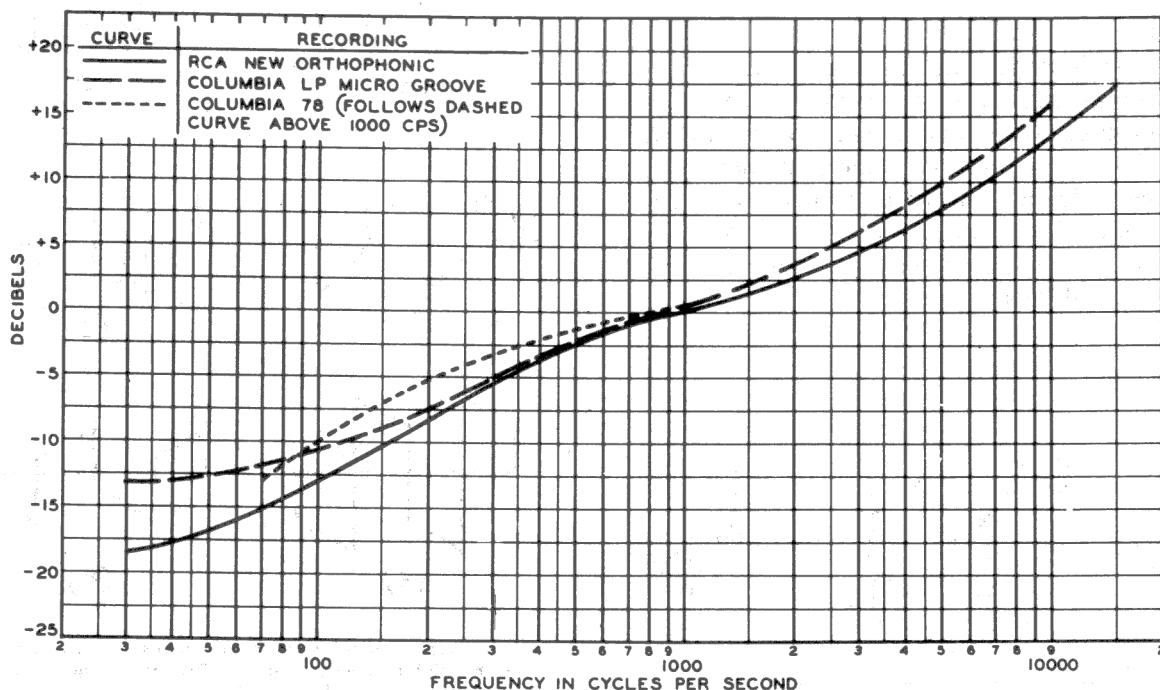


FIGURE 49

The recording curve is used as the basis for designing equalizer networks. A properly designed equalizer in an ideal amplifier will have an output which is the inverse of the recording curve. In other words, if the recording attenuates frequencies below 500 cps at a certain rate, the equalized amplifier must boost these frequencies at the same rate.

The WA-44A may be used to determine the response of an equalizer by feeding a low-frequency signal into the input and plotting the output in db (see Figure 46) as the frequency is increased to the cross-over point. Comparison with the recording curve for which the equalizer is designed will show the points of variation.

**Input and Output Impedances** — The input and output impedance of amplifiers and similar circuits may be quickly determined by means of a signal from the WA-44A and a vtvm. This method is accurate if the impedance to be measured is resistive, approximate if it is reactive.

Since most amplifiers have inputs which are mostly resistive, it is sufficient to check the input impedance with a low-frequency signal. A frequency of 50 cps may be used when checking a high-fidelity amplifier. For checking the input impedance of a PA system, or an amplifier with limited low-frequency response, a test frequency of 100 cps may be used. The circuit shown in Figure 50 is used to determine both high- and low-input impedances. For high-impedance circuits use the HI connector, for low-impedance circuits use the LO connector.  $R_i$  is used for input impedance measurements, while  $R_o$  is used for output impedance measurements.

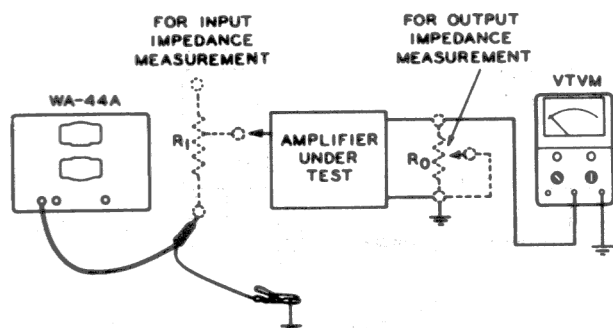


FIGURE 50

Set the frequency of the WA-44A to either 50 or 100 cps depending on the quality of the equipment under test. Connect the WA-44A to the amplifier; set the OUTPUT control so that there is a convenient reading on the vtvm scale. Keep the signal level low so that the amplifier is not overloaded.

Connect the potentiometer ( $R_i$ ) into the circuit and vary its resistance until the voltage indicated on the vtvm is one-half the original value. The resistance of the potentiometer is then equal to the input impedance.

Output impedances are measured by the same technique except that the potentiometer ( $R_o$ ) is shunted across the output of the amplifier. Since the output impedance changes as the frequency is varied, it is advisable to check the output impedance over a wide band of frequencies.

**Intermodulation Distortion** — Intermodulation distortion is due to the interaction of two or more frequencies in a non-linear amplifier. This interaction results in the modulation of one frequency by the other, thereby producing spurious frequencies. The extent of this type of distortion is expressed as a percentage, and is, in most cases, a better indication of the quality of an amplifier than is a measurement of total harmonic distortion.

The simplest and most direct method of measuring intermodulation distortion is the one developed by the Society of Motion Picture and Television Engineers. This method consists of simultaneously feeding a low-frequency and a high-frequency signal into the amplifier under test; the low frequency signal having an amplitude four times that of the high-frequency signal. The modulation of the high frequency signal is measured directly with an intermodulation distortion meter, or it can be determined by the use of an oscilloscope and a suitable filter.

The WA-44A provides a 60 cps signal from the LINE FREQ. terminal which, in conjunction with a higher frequency signal from the output terminal of the WA-44A, can be used for intermodulation tests. The equipment set-up is shown in Figure 51.

Set the WA-44A so that the output of the LO connector is 2 volts at 3000 cps. Connect a lead to the LINE FREQ. terminal and

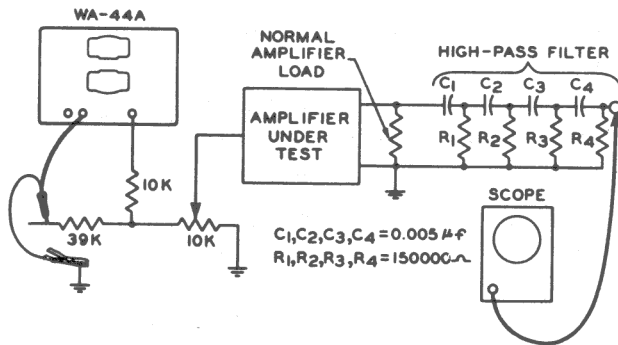


FIGURE 51

adjust the LINE FREQ. OUTPUT control so that 2 volts are available at the terminal. The attenuator consisting of a 10,000-ohm resistor and a 39,000-ohm resistor is used to provide the proper voltage relationship at the input of the amplifier.

The function of the high-pass filter is to bypass the low frequency signal so that only the high frequency signal appears at the vertical input of the oscilloscope.

If intermodulation distortion is present the 3000 cps signal will be modulated by the 60 cps signal, and the trace will appear as shown in Figure 52. The percentage of modulation distortion may be determined by use of the formula

$$\% \text{ I D } = \frac{100 (B-A)}{A}$$

where A = peak value of the unmodulated signal

B = peak value of the modulated signal

**Resonant Frequency of Speakers**—It is often important to know the resonant frequency of a speaker. This is especially so in the designing or the tuning of a speaker enclosure. The test setup shown in Figure 53 may be used to determine the resonant frequency of speakers. The speaker should be held in open air away from any large surface which might act as a load and change the resonant frequency. At resonance the voltmeter will indicate a voltage peak. In most speakers the resonant frequency is in the range from 50 to 250 cps.

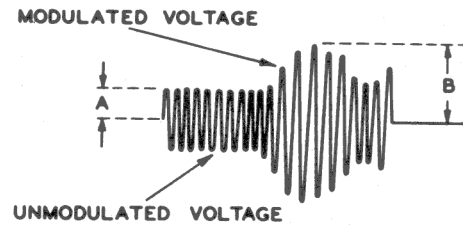


FIGURE 52

**Tuning a Bass-reflex Enclosure** — The test setup shown in Figure 53 can also be used to determine the proper port opening and enclosure volume necessary to produce the best possible low-frequency response. Excessive enclosure volume and too large a port will produce “boom.” On the other hand, a port of insufficient area and an enclosure which is too small will not extend the low-frequency response of the speaker.

A properly tuned enclosure will show two voltage rises, one above and the other below the resonant frequency of the speaker. The size of the port and the volume of the enclosure should be adjusted so that these voltage rises are as small as possible and of equal amplitude.

In cases where the voltmeter does not give an adequate reading, an oscilloscope should be used as an indicating device. A line-to-voice-coil transformer may also be used to effect a proper match.

If the amplifier which will be used with the speaker under test is available, connect the

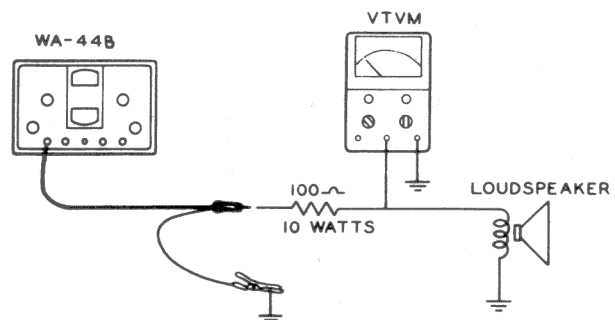


FIGURE 53

WA-44A to the amplifier input, and the speaker to be tested to the amplifier output, and use the voltmeter as indicated above.

**Locating Speaker and Cabinet Rattles —**

Vibrations and rattles which are sometimes produced in cabinets and speakers when certain audio frequencies are reproduced, may be found by feeding the output of the WA-44A into the audio section of a receiver or

amplifier, and varying the frequency of the generator from 11 cps through the audible range, until the vibration is heard. Cabinet vibrations are easily found by damping the section of the cabinet with the hand until the vibration stops or is diminished. Once the section is located, regluing, or bracing the section will usually correct the trouble. Speaker rattles require repair of the defective part or reconing.

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